



## **Avaya Solution & Interoperability Test Lab**

---

# **Application Notes for Configuring Windstream SIP Trunk Service with Avaya Communication Server 1000 Release 7.6, and Avaya Session Border Controller for Enterprise Release 6.2.1 – Issue 1.0**

## **Abstract**

These Application Notes describe the procedure for configuration Windstream SIP Trunk Service with Avaya Communication Server 1000 Release 7.6, and Avaya Session Border Controller for Enterprise Release 6.2.1.

The test was performed to verify SIP trunk features including basic calls, call forward (all calls, busy, no answer), call transfer (blind and consult), conference, and voice mail. Calls were placed to and from the PSTN with various Avaya endpoints.

Windstream SIP Trunk Service provides PSTN access via SIP trunks between enterprise and Windstream's network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

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.

## Table of Contents

1.	Introduction.....	5
2.	General Test Approach and Test Results.....	5
2.1.	Interoperability Compliance Testing.....	5
2.2.	Test Results .....	6
2.3.	Support .....	7
3.	Reference Configuration.....	8
4.	Equipment and Software Validated .....	9
5.	Configure Avaya Communication Server 1000.....	11
5.1.	Log into Communication Server 1000 System .....	11
5.1.1.	Log into System Manager and Element Manager (EM) .....	11
5.1.2.	Log into the Call Server by using the Overlay Command Line Interface (CLI) ....	13
5.2.	Administer an IP Telephony Node.....	14
5.2.1.	Obtain Node IP address .....	14
5.2.2.	Administer Terminal Proxy Server (TPS) .....	16
5.2.3.	Administer Quality of Service (QoS) .....	17
5.2.4.	Synchronize New Configuration.....	17
5.3.	Administer Voice Codec .....	18
5.3.1.	Enable Voice Codec G.711, G.729.....	18
5.3.2.	Enable Voice Codec on Media Gateways.....	19
5.4.	Zones and Bandwidth Management.....	20
5.4.1.	Create a Zone for IP Phones (Zone 10) .....	20
5.4.2.	Create a Zone for Virtual SIP Trunk (Zone 255).....	21
5.5.	Administer SIP Trunk Gateway .....	22
5.5.1.	Integrated Services Digital Network (ISDN).....	22
5.5.2.	Administer SIP Trunk Gateway to Avaya Communication Server 1000 .....	23
5.5.3.	Administer Virtual D-Channel.....	25
5.5.4.	Administer Virtual Super-Loop .....	29
5.5.5.	Administer Virtual SIP Routes .....	29
5.5.6.	Administer Virtual Trunks .....	31
5.5.7.	Administer Calling Line Identification Entries.....	34
5.5.8.	Enable External Trunk to Trunk Transfer.....	36
5.6.	Administer Dialing Plans .....	37
5.6.1.	Define ESN Access Codes and Parameters (ESN) .....	37
5.6.2.	Associate NPA and SPN call to ESN Access Code 1.....	38
5.6.3.	Digit Manipulation Block Index (DMI).....	39

5.6.4.	Route List Block (RLB) (RLB 14) .....	40
5.6.5.	Inbound Call – Incoming Digit Translation Configuration .....	42
5.6.6.	Outbound Call - Special Number Configuration .....	44
5.6.7.	Outbound Call - Numbering Plan Area (NPA).....	45
5.7.	Administer a Phone .....	46
5.7.1.	Phone creation.....	46
5.7.2.	Enable Privacy for the Phone.....	47
5.7.3.	Enable Call Forward for Phone.....	48
5.7.4.	Enable Call Waiting for Phone .....	50
6.	Configure Session Border Controller for Enterprise .....	51
6.1.	Log into the Avaya SBCE.....	51
6.2.	Global Profiles.....	52
6.2.1.	Configure Server Interworking - Avaya site.....	52
6.2.2.	Configure Server Interworking – Windstream site .....	53
6.2.3.	Configure URI Groups.....	54
6.2.4.	Configure Routing – Avaya site .....	54
6.2.5.	Configure Routing – Windstream site .....	55
6.2.6.	Configure Signaling Manipulation .....	56
6.2.7.	Configure Server – Communication Server 1000.....	57
6.2.8.	Configure Server – Windstream .....	58
6.2.9.	Configure Topology Hiding – Avaya site.....	60
6.2.10.	Configure Topology Hiding – Windstream site .....	61
6.3.	Domain Policies .....	61
6.3.1.	Create Application Rules .....	62
6.3.2.	Create Border Rules .....	63
6.3.3.	Create Media Rules.....	64
6.3.4.	Create Security Rules.....	65
6.3.5.	Create Signaling Rules.....	66
6.3.6.	Create Time of Day Rules.....	68
6.3.7.	Create Endpoint Policy Groups .....	70
6.3.8.	Create Session Policy.....	72
6.4.	Device Specific Settings.....	73
6.4.1.	Manage Network Settings.....	73
6.4.2.	Create Media Interfaces .....	74
6.4.3.	Create Signaling Interfaces .....	75

6.4.4.	Configuration Server Flows .....	75
6.4.5.	Create Session Flows .....	77
7.	Windstream SIP Trunking Service Configuration .....	78
8.	Verification Steps.....	78
8.1.	General .....	78
8.2.	Verification of an Active Call on Communication Server 1000.....	78
8.3.	Protocol Trace .....	81
9.	Conclusion .....	82
10.	References.....	83
11.	Appendix A: SigMa Script.....	84

# 1. Introduction

These Application Notes illustrate a sample configuration using Avaya Communication Server 1000 Release 7.6, Avaya Session Border Controller for Enterprise (Avaya SBCE) Release 6.2.1 with Windstream SIP Trunk Service using Broadsoft switch. Windstream SIP Trunk Service provides PSTN access via SIP Trunks between the enterprise and Windstream's network as an alternative to legacy analog or digital trunks.

## 2. General Test Approach and Test Results

Communication Server 1000 was connected to Avaya SBCE via SIP Trunks. Avaya SBCE was connected to Windstream's network via SIP trunks. Various call types were made from Communication Server 1000 to Windstream and vice versa to verify interoperability.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution

### 2.1. Interoperability Compliance Testing

Compliance testing scenarios for the configuration described in these Application Notes included the following:

- General call processing between Communication Server 1000 and Windstream SIP Trunk Service, including the following:
  - Codec/ptime (G.729/20ms, and G.711 u-law/20ms), no Voice Activity Detection (VAD).
  - Hold/Resume on both ends.
  - Calling Line Identification Display (CLID).
  - Ring-back tone.
  - Speech (audio) path.
  - Dialing plan support (local, long distance, international, outbound toll-free, assisted operator, 411, and 911).
  - Advanced features (Call on Mute, Call Park, and 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.
- Response to SIP OPTIONS queries.
- Fax G.711 Pass Through.
- Inbound and outbound long hold time call stability.
- Caller number/ID presentation.

- Privacy requests (i.e., caller anonymity) and Caller ID restriction for inbound and outbound calls.
- DTMF (RFC2833) in both directions.
- SIP Transport UDP, port 5060.
- Voice Mail Server Call Pilot (hosted on Avaya system).
- Off-net call forwarding (Windstream supports Diversion Header).

The following assumptions were made for the compliance tested configuration:

1. Communication Server 1000 R7.6 software with latest patches.
2. Windstream Service provides support to setup, configure and troubleshoot on carrier switch during testing execution.

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 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 was open during the test cases execution for the monitoring of BUG(s), ERROR and AUD messages (See **Session 5.1.2**).
8. Speech path was checked before and after calls were put on/off hold from each end.
9. Calls were checked to ensure that all resources such as Virtual trunks, TDM trunks, Sets and VGWs (Voice Gateways) were released when calls were ended (See **Session 8.2** - SIP Trunk monitoring (LD 32)).

## 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:

1. If the Avaya Communication Server 1000 phone holds/resumes an outbound call, the dialed digits are no longer displayed. This is a Communication Server 1000 known issue.
2. Windstream Service does not support the inbound-toll-free call. This function is not supported in Windstream Lab environment, but it is supported in Windstream production environment.
3. CLID (Calling Line Identification Display) and CPND (Calling Party Name Display) are blocked intermittently in international inbound calls.
4. Call Forward to international numbers does not work properly - Windstream Service supports Call Forward to any numbers; US or international. But in this compliance testing, Windstream Service only supports Call Forward to US numbers. Call Forward to international numbers is not allowed because the route to the international test carrier is not working properly in current Windstream Lab environment.

5. PSTN1 phone calls to Avaya Communication Server 1000 phone, then the Avaya Communication Server 1000 phone does a blind transfer to the PSTN2 phone. The PSTN1 phone could not hear ring-back-tone from the PSTN2 phone when the Avaya Communication Server 1000 phone completed the blind transfer. In order to resolve this ring-back-tone issue in this particular scenario, Avaya SBCE is configured to translate the SIP 180 with SDP to SIP 180 without SDP (see **Sections 6.2.1**) so that PSTN1 can hear the local ring-back-tone. However, if this translation is performed on Avaya SBCE, early media is not supported in this configuration. Please communicate this requirement to Windstream Service before implementing the translation on the Avaya SBCE.
6. Windstream Service does not support registration and authentication.

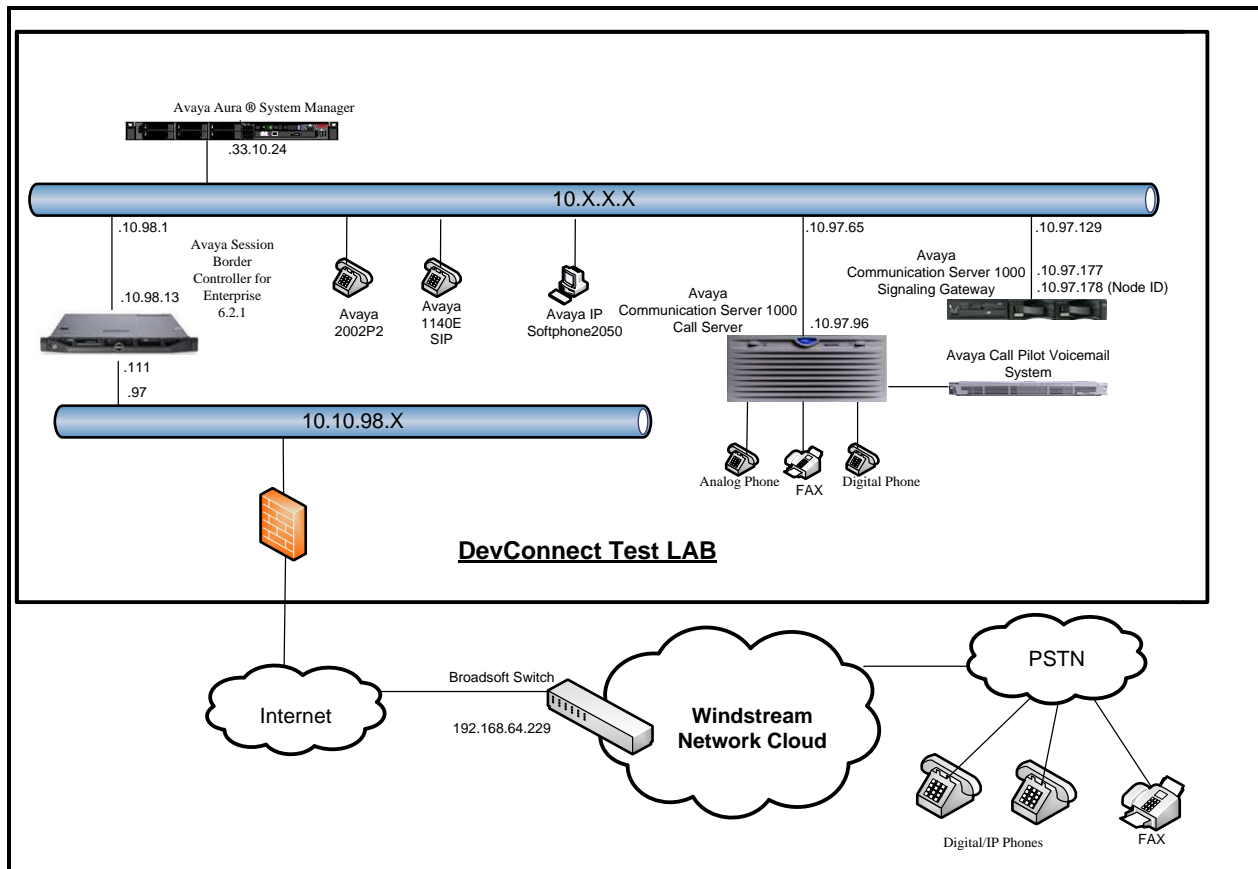
## 2.3. Support

For technical support on the Avaya products described in these Application Notes visit:  
<http://support.avaya.com>.

For technical support on the Windstream Service, please contact customer service or visit  
<http://www.windstream.com/>

### 3. Reference Configuration

**Figure 1** illustrates the test configuration used during the compliance test between Communication Server 1000 and Windstream Service. For confidentiality and privacy purposes, actual public IP addresses used in this testing have been masked and replaced with fictitious IP addresses throughout the document.



**Figure 1- Network diagram for Avaya and Windstream SIP Trunk Service**



## 4. Equipment and Software Validated

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

### Avaya systems:

Equipment/Software	Release/Version
Avaya Communication Server 1000 (CPPM)	Call Server: 765 P + Signaling Server: 7.65.16 GA SIP Line Server: 7.65.16 GA
Avaya S8800 Server	Avaya Aura @System Manager R6.3.4 – FP3 6.3.0.8.5682 – 6.3.8.2631
Avaya Call Pilot C201i	Call Pilot Voice Mail Manager: 05.00.41.143
Avaya Session Border Controller for Enterprise	6.2.1 Q07
Avaya Phones: 2002 p2 1140E SIP	0604DCO 4.3.12.0
Avaya 3904 Digital Phone	N/A
Avaya 2050 IP Softphone	4.04.0067
Analog Symphony 2000	N/A
HP Office jet 4500 Fax	N/A

### Windstream Service:

System	Software
Broadsoft	17sp4
Acme Packet Net 4250	SC6.2.0 Patch 3

Additional patch lineup for the configuration listed as follows:

**Call Server:** 7.65 P+ GA plus latest DEPLIST – CPL\_7.6S4.zip (X2107.65P)

**Signaling Server:** 7.65.16 GA plus latest DEPLIST – SP\_7.6\_4.ntl (7.65.16.00)

CS1K Signaling Server patch list:

```
[admin@car3-ssg-carrier ~]$ pstat
```

```
Product Release: 7.65.16.00
```

```
In system patches: 1
```

```
PATCH# NAME IN_SERVICE DATE SPECINS TYPE RPM
```

```
37 p31484_1 Yes 20/02/14 NO FRU cs1000-shared-general-7.65.16-00.i386
```

```
In System service updates: 26
```

```
PATCH# IN_SERVICE DATE SPECINS REMOVABLE NAME
```

```
9 Yes 20/02/14 YES YES cs1000-dmWeb-7.65.16.22-1.i386.000
```

```
12 Yes 19/02/14 NO YES cs1000-linuxbase-7.65.16.22-02.i386.000
```

```
13 Yes 20/02/14 NO YES cs1000-pd-7.65.16.21-00.i386.000
```

```
14 Yes 20/02/14 NO YES cs1000-Jboss-Quantum-7.65.16.22-3.i386.000
```

```
15 Yes 20/02/14 YES YES cs1000-patchWeb-7.65.16.22-1.i386.000
```

```
16 Yes 20/02/14 NO YES cs1000-shared-carrdtct-7.65.16.21-01.i386.000
```

```
17 Yes 20/02/14 NO YES cs1000-shared-tpselect-7.65.16.21-01.i386.000
```

```
18 Yes 20/02/14 NO YES cs1000-dbcom-7.65.16.21-00.i386.000
```

19	Yes	20/02/14	NO	YES	cs1000-shared-xmsg-7.65.16.21-00.i386.000
20	Yes	20/02/14	NO	YES	cs1000-mscAnnc-7.65.16.21-02.i386.001
21	Yes	20/02/14	NO	YES	cs1000-mscAttn-7.65.16.21-04.i386.001
22	Yes	20/02/14	NO	YES	cs1000-mscConf-7.65.16.21-02.i386.001
23	Yes	20/02/14	NO	YES	cs1000-mscMusc-7.65.16.21-02.i386.001
24	Yes	20/02/14	NO	YES	cs1000-mscTone-7.65.16.21-03.i386.001
25	Yes	20/02/14	NO	YES	cs1000-gk-7.65.16.21-01.i386.000
26	Yes	20/02/14	NO	YES	cs1000-snmp-7.65.16.21-00.i686.000
27	Yes	20/02/14	YES	YES	tzdata-2013c-2.el5.i386.001
28	Yes	20/02/14	YES	YES	cs1000-tps-7.65.16.21-11.i386.000
29	Yes	20/02/14	NO	YES	cs1000-sps-7.65.16.21-8.i386.000
30	Yes	20/02/14	NO	YES	cs1000-shared-omm-7.65.16.21-2.i386.000
31	Yes	20/02/14	YES	YES	cs1000-baseWeb-7.65.16.22-1.i386.000
32	Yes	20/02/14	YES	YES	cs1000-csoneksvrmgr-7.65.16.22-1.i386.000
33	Yes	20/02/14	YES	YES	cs1000-ipsec-7.65.16.22-1.i386.000
34	Yes	20/02/14	YES	YES	cs1000-vtrk-7.65.16.22-4.i386.000
35	Yes	20/02/14	NO	YES	cs1000-cppmUtil-7.65.16.22-1.i686.000
36	Yes	20/02/14	YES	YES	cs1000-oam-logging-7.65.16.22-3.i386.000

## 5. Configure Avaya Communication Server 1000

These Application Notes use the Incoming Digit Translation feature to receive calls, the Numbering Plan Area Code (NPA), and Special Number (SPN) features to route calls from the Communication Server 1000 to the PSTN, via SIP trunks to Windstream.

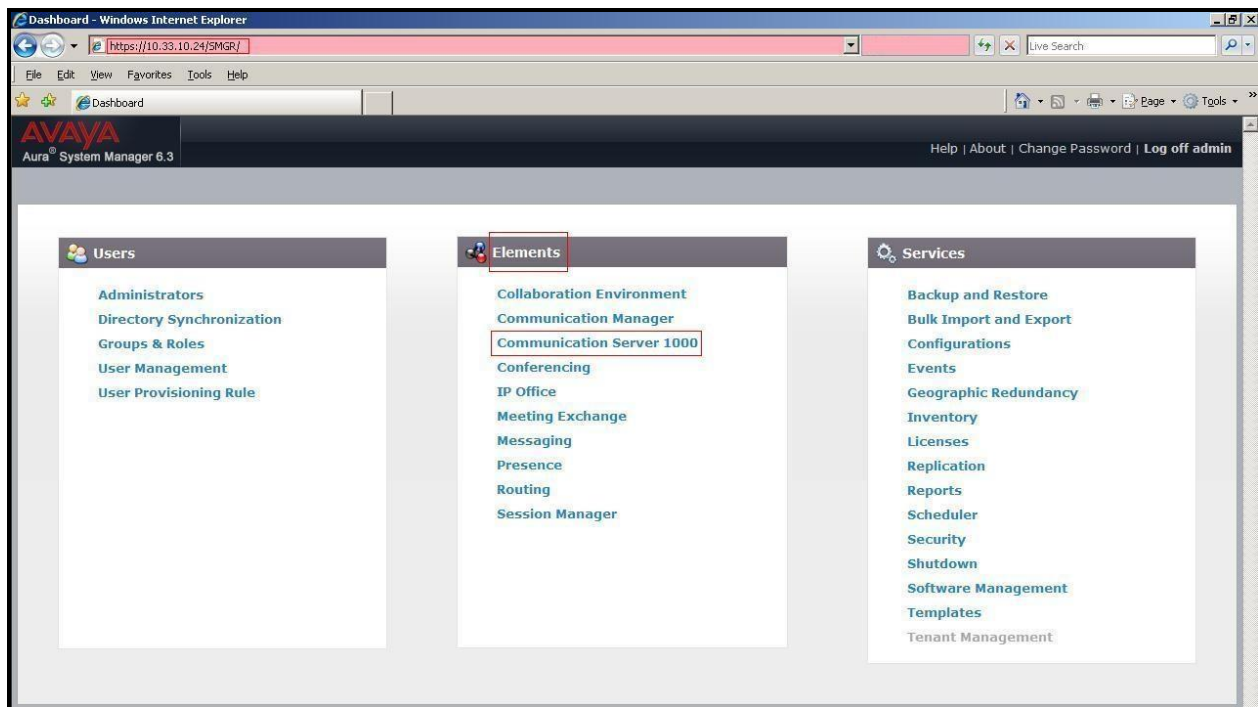
These Application Notes assume that the basic Communications Server 1000 configuration has already been administered. For further information on Communications Server 1000, please consult the references in **Section 10**.

The procedures below describe the configuration details for configuring the Communication Server 1000.

### 5.1. Log into Communication Server 1000 System

#### 5.1.1. Log into System Manager and Element Manager (EM)

Open an instance of a web browser and connect to the System Manager using the following address: `https://<System Manager IP address>/SMGR/`. Log in using an appropriate User ID and Password (not shown). Select **Elements** → **Communication Server 1000**



**Figure 2 –System Manager Home Screen**

The **Avaya Communication Server 1000 Management** screen is displayed. Click on the **Element Name** of the Communication Server 1000 Element as highlighted in red box as below:

The screenshot shows the Avaya Aura System Manager 6.3 interface. The left sidebar contains a navigation tree with categories like Network, User Services, and Security. The main content area is titled 'Elements' and displays a table of system elements. The element 'EM on car3-sipl-ucm' is highlighted with a red box. Below the table, there are buttons for 'Add...', 'Edit...', and 'Delete'.

Element Name	Element Type	Release	Address	Description
smgr.bvdev.com (primary)	Base OS	7.6	10.33.10.24	Base OS element
EM on car3-sipl-ucm	CS1000	7.6	10.10.97.96	New element
car3-cores.bvdev.com (member)	Linux Base	7.6	10.10.97.179	Base OS element
car3-sipl-ucm.bvdev.com (member)	Linux Base	7.6	10.10.97.175	Base OS element
car3-ssq-carrier.bvdev.com (member)	Linux Base	7.6	10.10.97.177	Base OS element
10.10.97.97	Media Gateway Controller	7.6	10.10.97.97	New element

**Figure 3 – Communication Server 1000 Management**

Log into the Communication Server 1000 using an appropriate **User ID** and **Password**.

The screenshot shows the Avaya Communication Server 1000 Log In screen. It features a red header with the Avaya logo. Below the header, there is a login form with fields for 'User ID' (containing 'admin') and 'Password' (masked with dots). A 'Log In' button is present, along with a 'Change Password' link. A note at the bottom left states: 'Go to central login for Single Sign-On'.

**Figure 4 – Communication Server 1000 Log In Screen**

The Communication Server 1000 Element Manager **System Overview** page is displayed as shown in **Figure 5**.

IP Address: 10.10.97.96

Type: Avaya Communication Server 1000E CPPM Linux

Version: 4121

Release: 765 P +



**Figure 5 – Element Manager System Overview**

### 5.1.2. Log into the Call Server by using the Overlay Command Line Interface (CLI)

Using Putty, SSH to the IP address of the Communication Server 1000 Signaling Server using an account with administrator credentials.

Run the command **cslogin** and log in with the appropriate user account and password. Sample output is shown below.

login as: < --- **enter an account with administrator credentials**

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@10.10.97.177's password: <----**enter the password**

Last login: Mon Feb 24 07:20:18 2014 from 10.10.98.78

[admin@car3-ssg-carrier ~]\$ **cslogin**

SEC054 A device has connected to, or disconnected from, a pseudo tty without authenticating  
>login

USERID? < --- **enter the user account**

PASS? <----**enter the password**

TTY #08 LOGGED IN ADMIN 07:39 24/02/2014

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 log out immediately. This system may be monitored for operational purposes at any time.

>

**Note:** Leave this screen on for monitoring of BUG(s), ERROR and AUD messages.

## 5.2. Administer an IP Telephony Node

This section describes how to configure an IP Telephony Node on Communication Server 1000.

### 5.2.1. Obtain Node IP address

These Application Notes assume that the basic Communication Server 1000 configuration has already been administered and that a Node has already been created. This section describes the steps for configuring a Node (Node ID 3000) in Communication Server 1000 IP network to work with Windstream SIP Trunk Service. For further information on Communications Server 1000, please consult the references in **Section 10**.

Select **System** → **IP Network** → **Nodes: Servers, Media Cards** and then click on the **Node ID** as shown in **Figure 6**.

The screenshot displays the AVAYA CS1000 Element Manager web interface. On the left is a navigation tree with categories like UCM Network Services, Home, Links, System, and IP Network. The 'IP Network' section is expanded, showing 'Nodes: Servers, Media Cards' highlighted. The main content area is titled 'IP Telephony Nodes' and shows a table of nodes. The table has columns for Node ID, Components, Enabled Applications, ELAN IP, Node/TLAN IPv4, Node/TLAN IPv6, and Status. Two nodes are listed: Node ID 3000 and Node ID 3002, both with a status of 'Synchronized'. Above the table are buttons for Add, Import, Export, and Delete. Below the table are checkboxes for 'Nodes', 'Component servers and cards', and 'IPv6 address'.

Node ID	Components	Enabled Applications	ELAN IP	Node/TLAN IPv4	Node/TLAN IPv6	Status
3000	1	LTPS, Gateway ( SIPGw )	-	10.10.97.178		Synchronized
3002	1	SIP Line, LTPS	-	10.10.97.176		Synchronized

**Figure 6 – IP Telephony Nodes**

The **Node Details** screen is displayed in **Figure 7** with the IP address of the Communication Server 1000 node. **Call server IP address: 10.10.97.96**. The **Node IPv4 address 10.10.97.178** is a virtual address which corresponds to the **TLAN IP address 10.10.97.177** of the Signaling Server/SIP Signaling Gateway. The SIP Signaling Gateway uses this Node IP address to communicate with other components to process SIP calls.

**AVAYA CS1000 Element Manager**

Managing: 10.10.97.96 Username: admin  
System » IP Network » IP Telephony Nodes » Node Details

**Node Details (ID: 3000 - LTPS, Gateway ( SIPGw ))**

Node ID:  \* (0-9999)

Call server IP address:  \*

TLAN address type: ☒ IPv4 only  
☐ IPv4 and IPv6

**Embedded LAN (ELAN)**

Gateway IP address:  \*

Subnet mask:  \*

**Telephony LAN (TLAN)**

Node IPv4 address:  \*

Subnet mask:  \*

Node IPv6 address:

\* Required Value. Save Cancel

**Associated Signaling Servers & Cards**

Select to add Add Remove Make Leader Print | Refresh

Hostname	Type	Deployed Applications	ELAN IP	TLAN IPv4	Role
<input type="checkbox"/> car3-ssg-carrier	Signaling_Server	SIP Line, LTPS, Gateway (SIP/H323), PD, Presence Publisher, IP Media Services	10.10.97.95	10.10.97.177	Leader

Show: ☐ IPv6 address

Note: Only server(s) that are not part of any other IP telephony node and deployed application(s) that match the service(s) selected for this node are available in the servers list.

**Figure 7 –Node Details 1**



The **Node Details** screen is displayed in **Figure 8** with the IP Telephony Node Properties and Applications.

**AVAYA CS1000 Element Manager**

Managing: 10.10.97.96 Username: admin  
System » IP Network » IP Telephony Nodes » Node Details

**Node Details (ID: 3000 - LTPS, Gateway (SIPGw))**

Subnet mask: 255.255.255.192 \*      Subnet mask: 255.255.255.192 \*  
Node IPv6 address: [ ]

**IP Telephony Node Properties**

- Voice Gateway (VGW) and Codecs
- Quality of Service (QoS)
- LAN
- SNTIP
- Numbering Zones
- MCDN Alternative Routing Treatment (MALT) Causes

**Applications (click to edit configuration)**

- SIP Line
- Terminal Proxy Server (TPS)
- Gateway (SIPGw)
- Personal Directories (PD)
- Presence Publisher
- IP Media Services

\* Required Value.      **Save**      **Cancel**

**Associated Signaling Servers & Cards**

Select to add      **Add**      **Remove**      **Make Leader**      **Print** | **Refresh**

<input type="checkbox"/> Hostname	Type	Deployed Applications	ELAN IP	TLAN IPv4	Role
<input type="checkbox"/> car3-ssg-carrier	Signaling_Server	SIP Line, LTPS, Gateway (SIP/H323), PD, Presence Publisher, IP Media Services	10.10.97.95	10.10.97.177	Leader

Show: ☐ IPv6 address

**Figure 8 –Node Details 2**

### 5.2.2. Administer Terminal Proxy Server (TPS)

Continuing from **Section 5.2.1**, on the **Node Details** page, select the **Terminal Proxy Server (TPS)** link as shown in **Figure 8**. Check the **UNISim Line Terminal Proxy Server** checkbox to enable proxy service on this node and then click the **Save** button as shown in **Figure 9**.

**AVAYA CS1000 Element Manager**

Managing: 10.10.97.96 Username: admin  
System » IP Network » IP Telephony Nodes » Node Details » UNISim Line Terminal Proxy Server (LTPS) Configuration

**Node ID: 3000 - UNISim Line Terminal Proxy Server (LTPS) Configuration Details**

**Firmware | DTLS | Network Connect Server**

UNISim Line Terminal Proxy Server: ☒ Enable proxy service on this node

**Firmware**

IP address: 0.0.0.0  
Full file path: download/firmware  
Server Account/User ID: [ ]  
Password: [ ]

**DTLS**

DTLS policy: Off

Options: ☐ Client authentication  
☐ Periodic re-keying

**Network Connect Server**

\* Required Value.      **Save**      **Cancel**

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

**Figure 9 – TPS Configuration Details**



### 5.2.3. Administer Quality of Service (QoS)

Continuing from **Section 5.2.1**, on the **Node Details** page, select the **Quality of Service (QoS)** link as shown in **Figure 8**. The default Diffserv values are as shown in **Figure 10**. Click on the **Save** button.

The screenshot shows the CS1000 Element Manager interface. The top bar includes the Avaya logo, the title 'CS1000 Element Manager', and links for 'Help' and 'Logout'. The breadcrumb trail indicates the path: 'System > IP Network > IP Telephony > Nodes > Node Details > Quality of Service (QoS)'. The main content area is titled 'Node ID: 3000 - Quality of Service (QoS)'. It contains a 'Diffserv Codepoint (DSCP)' configuration section with the following fields:

- 'Enable Avaya automatic QoS' checkbox (unchecked).
- 'Control packets' input field with value '40' and range '(0-63)'.
- 'Voice packets' input field with value '40' and range '(0-63)'.
- 'VLAN tagging' checkbox (unchecked) and '802.1Q support' checkbox (checked).
- '802.1Q bits value (802.1P)' input field with value '5' and range '(0-7)'.

At the bottom of the configuration section, there is a note: 'Note: Changes made on this page will NOT be transmitted until the Node is also saved.' Below the note are 'Save' and 'Cancel' buttons. The sidebar on the left contains various navigation links, including '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', and 'Dialing and Numbering Plans'.

**Figure 10 – QoS Configuration Details**

### 5.2.4. Synchronize New Configuration

Continuing from **Section 5.2.3**, return to the **Node Details** page (**Figure 7**) and click on the **Save** button. The **Node Saved** screen is displayed. Click on **Transfer Now** (not shown). The **Synchronize Configuration Files (Node ID <3000>)** screen is displayed (not shown). Check the **Signaling Server** checkbox and click on **Start Sync** (not shown). When the synchronization completes, check the **Signaling Server** checkbox and click on the **Restart Applications** (not shown).

## 5.3. Administer Voice Codec

### 5.3.1. Enable Voice Codec G.711, G.729

Select **IP Network** → **Nodes: Servers, Media Cards** from the left pane and on the **IP Telephony Nodes** screen displayed (not shown), select the **Node ID** of the Communication Server 1000 system. The **Node Details** screen is displayed, (see **Section 5.2.1** for more details). On the **Node Details** page shown in **Figure 8**, click on **Voice Gateway (VGW) and Codecs**. Windstream supports **G.711/time 20ms** and **G.729/time 20ms** with **Voice Activity Detection (VAD)** checkbox unchecked. Click on the **Save** button.

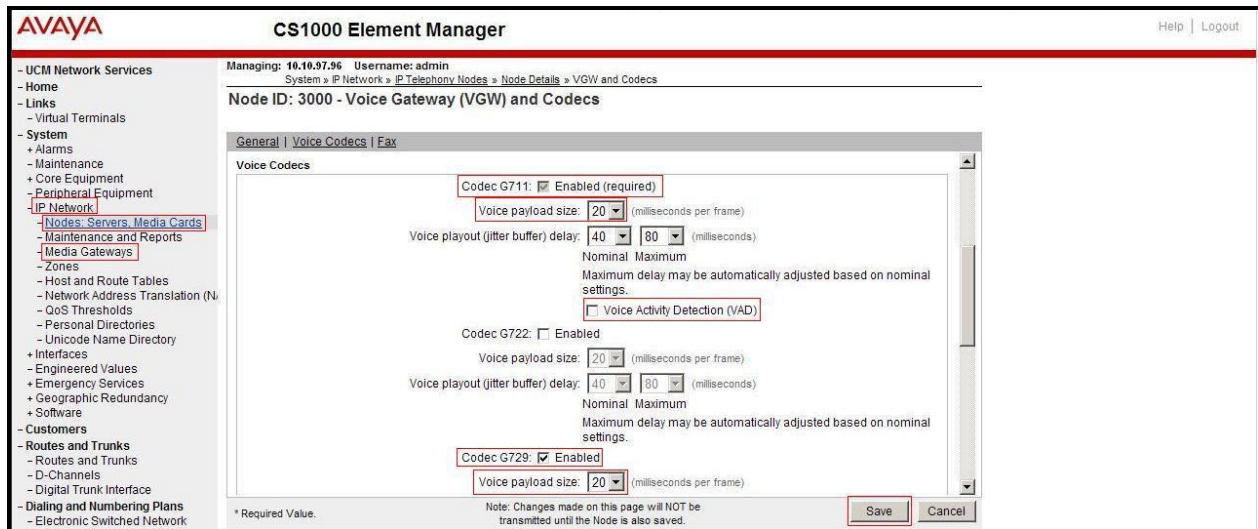


Figure 11 – Voice Gateway and Codec Configuration Details

Synchronize the new configuration (please refer to **Section 5.2.4**).

### 5.3.2. Enable Voice Codec on Media Gateways

From the left menu of the Element Manager page in **Figure 11**, select **IP Network → Media Gateways**. The Media Gateways page will appear (not shown). Click on the **MGC** which is located on the right of the page. In the following screen, scroll down to select the **Codec G.711** and **Codec G.729A** and uncheck **VAD** as shown in **Figure 12**. Scroll down to the bottom of the page and click on the **Save** button (not shown).

The screenshot displays the AVAYA CS1000 Element Manager interface. The left sidebar contains a navigation tree with categories like 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, 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, and Login Options. The main content area is titled 'VGW and IP phone codec profile' and contains the following settings:

- Enable echo canceller: ☒
- Echo canceller tail delay: 128 (milliseconds)
- Enable dynamic attenuation: ☒
- Voice activity detection threshold: 1 (0 - 4 DBM)
- Idle noise level: 0 (0 - 1 DBM)
- R factor calculation: ☐
- DTMF tone detection: ☒
- Enable low latency mode: ☐
- Remove DTMF delay (squell DTMF from TDM to IP): ☒
- Enable modem/fax pass through mode: ☒
- Enable V.21 FAX tone detection: ☒
- Fax TCF method: 2
- FAX maximum rate: 9600 (bps)
- 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 G729A: Select ☒
- Codec name: G729A
- Voice payload size: 20 (ms/frame)

Copyright © 2002-2011 Avaya Inc. All rights reserved.

**Figure 12 – Media Gateways Configuration Details**

## 5.4. Zones and Bandwidth Management

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

### 5.4.1. Create a Zone for IP Phones (Zone 10)

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

Select **IP Network** → **Zones** from the left pane (not shown), click on **Bandwidth Zones** as shown in **Figure 13**.

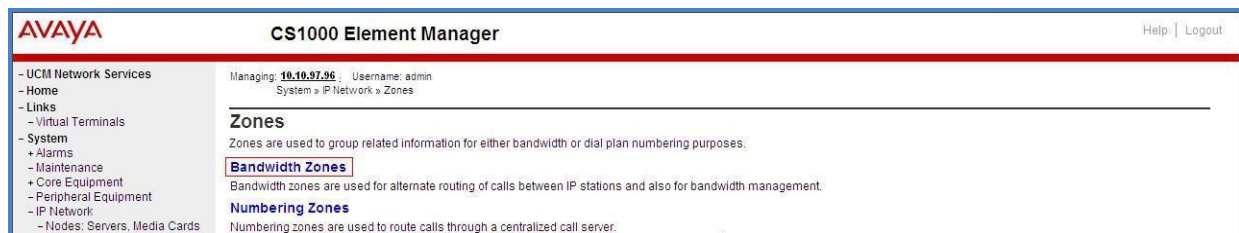


Figure 13 – Zones Page

The **Bandwidth Zones** screen is displayed as shown in **Figure 14**. Click **Add** to create a new zone for IP Phones.

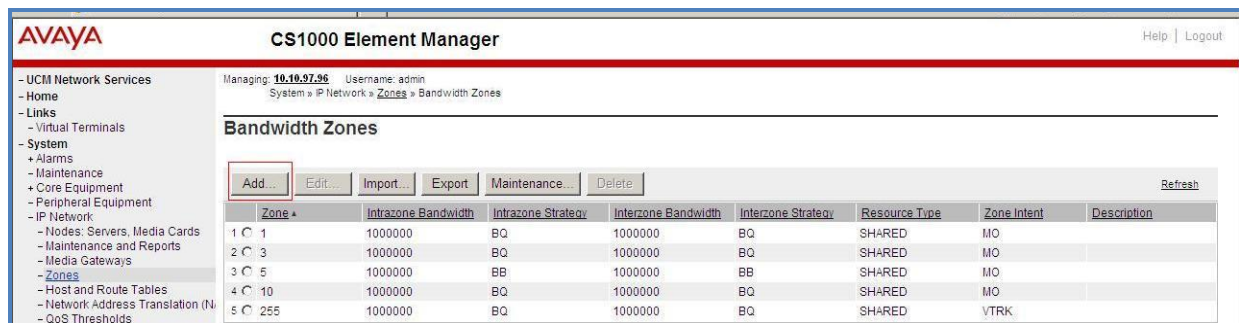


Figure 14 – Bandwidth Zones

Select and input the values as shown below (in the red boxes) in **Figure 15**, and click on the **Submit** button.

- **INTRA\_BW: 1000000**
- **INTRA\_STGY: Set codec for local calls. Select **Best Quality (BQ)** to use G.711 as the first priority codec for negotiation.**
- **INTER\_BW: 1000000**
- **INTER\_STGY: Set codec for the calls over trunk. Select **Best Quality (BQ)** to use G.711 as the first priority codec for negotiation.**
- **Zone Intent (ZBRN): Select **MO (MO)** for IP phones, and VGW.**

Managing: 10.10.97.96 Username: admin  
System » IP Network » Zones » Bandwidth Zones » Bandwidth Zones 10 » Edit Bandwidth Zone » 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):	

Submit Refresh Cancel

**Figure 15 –Bandwidth Management Configuration Details – IP phone**

### 5.4.2. Create a Zone for Virtual SIP Trunk (Zone 255)

Follow the steps described in **Section 5.4.1** to create a zone for the virtual SIP trunk. The difference is in the **Zone Intent (ZBRN)** field. Select **VTRK** for virtual trunk as shown in **Figure 16** and then click on the **Submit** button.

Managing: 10.10.97.96 Username: admin  
System » IP Network » Zones » Bandwidth Zones » Bandwidth Zones 255 » Edit Bandwidth Zone » 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):	

Submit Refresh Cancel

**Figure 16 –Bandwidth Management Configuration Details –virtual SIP trunk**

## 5.5. Administer SIP Trunk Gateway

This section describes the steps for establishing a SIP connection between the SIP Signaling Gateway and Avaya Session Border Controller.

### 5.5.1. Integrated Services Digital Network (ISDN)

Select **Customers** in the left pane (not shown). 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 **Customer 00 Edit** page. The screen is updated with a listing of available **Feature Packages** (not all features are shown in **Figure 17** below). Select **Integrated Services Digital Network** to edit the parameters shown below. Check the **Integrated Services Digital Network** option, and retain the default values for all remaining fields. Scroll down to the bottom of the screen, and click on the **Save** button (not shown).

**Figure 17 – Customer – ISDN Configuration**



## 5.5.2. Administer SIP Trunk Gateway to Avaya Communication Server 1000

Select **IP Network** → **Nodes: Servers, Media Cards** from the left pane. In the **IP Telephony Nodes** screen displayed (not shown), select the **Node ID** of the Communication Server 1000 system. The **Node Details** screen is displayed as shown in **Figure 8, Section 5.2.1**.

On the **Node Details** screen, select **Gateway (SIPGw)**. Under the **General** tab of the **Virtual Trunk Gateway Configuration Details** screen, 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 18**. The **SIP domain name** and **Local SIP port** should be matched in the configuration of Avaya SBCE (in **Sections 6.2.4, 6.2.7, and 6.2.9**).

**AVAYA CS1000 Element Manager**

Managing: 10.10.97.96 Username: admin  
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

**Node ID: 3000 - 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: bwdev7.com  
Local SIP port: 5060 \*(1 - 65535)  
Gateway endpoint name: car3-sng-carrier  
Gateway password:   
Application node ID: 3000 \*(0-9999)  
Enable failsafe NRS: ☐

Note: FailSafe NRS will be enabled only on those servers in the node where NRS application is not deployed.

**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 18 – Virtual Trunk Gateway Configuration Details**

Click on the **SIP Gateway Settings** tab, under **Proxy or Redirect Server**, and enter the following values (highlighted in red boxes) for the specified fields, retaining the default values for the remaining fields as shown in **Figure 19**. Enter the internal IP address of Avaya SBCE in the **Primary TLAN IP address** field. Enter **Port: 5060** and **Transport protocol: UDP**. Uncheck **Support registration** checkbox.

AVAYA CS1000 Element Manager

Managing: 10.10.97.96 Username: admin  
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

Node ID: 3000 - Virtual Trunk Gateway Configuration Details

General | **SIP Gateway Settings** | SIP Gateway Services

Proxy Or Redirect Server:

Proxy Server Route 1:

Primary TLAN IP address: 10.10.98.13  
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"

Port: 5060 (1 - 65535)

Transport protocol: UDP

Options: ☐ Support registration  
☐ Primary CDS proxy

Secondary TLAN IP address: 0.0.0.0  
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"

Port: 5060 (1 - 65535)

Transport protocol: TCP

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

Save Cancel

**Figure 19 – Virtual Trunk Gateway Configuration Details**

On the same page as shown in **Figure 19**, scroll down to the **SIP URI Map** section.

Under the **Public E.164 domain names**, enter the following:

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

Under the **Private domain names**, enter the following:

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



The remaining fields can be left at their default values as shown in **Figure 20**. Then click on the **Save** button.

**Figure 20 – Virtual Trunk Gateway Configuration Details**

Synchronize the new configuration (please refer to **Section 5.2.4**).

### 5.5.3. Administer Virtual D-Channel

Select **Routes and Trunks** → **D-Channels** (not shown) 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 and type **DCH** as shown in **Figure 21**. Click the **to Add** button.

**Figure 21 – D-Channels**

The **D-Channels 100 Property Configuration** screen is displayed next, as shown in **Figure 22**. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **D channel Card Type:** D-Channel is over IP (DCIP)
- **Designator:** A descriptive name
- **User:** Integrated Services Signaling Link Dedicated (ISLD)
- **Interface type for D-channel:** Meridian Meridian1 (SL1)
- **Meridian 1 node type:** Slave to the controller (USR)
- **Release ID of the switch at the far end:** 25

Click on **Advanced options (ADVOPT)**. Check on the **Network Attendant Service Allowed** checkbox as shown in **Figure 22**. Other fields are left as default.

**AVAYA CS1000 Element Manager**

**- Basic Configuration**

Input Description	Input Value
Action Device And Number (ADAN):	DCH
D channel Card Type:	DCIP
Designator:	VoIP
Recovery to Primary:	<input type="checkbox"/>
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:	<a href="#">more PRI</a>
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:	1800 (Range: 0 - 3700)
<b>+ Basic options (BSCOPT)</b>	
<b>- Advanced options (ADVOPT)</b>	
- Layer 3 call control message count per 5 second time interval:	300 (Range: 60 - 350)
- Number of Status Enquiry Messages sent within 128 ms:	1
- Map channel number to timeslots on a PRI2 loop:	<input checked="" type="checkbox"/>
<b>- H323 Overlap Signaling Settings (H323)</b>	
- Overlap Receiving:	<input type="checkbox"/>
- Overlap Sending:	<input type="checkbox"/>
- Overlap Timer:	
- Multilocution Business Group Allowed:	<input type="checkbox"/>
- Network Attendant Service Allowed:	<input checked="" type="checkbox"/>
<b>+ Link Access Protocol for D-channel (LAPD)</b>	
<b>+ Feature Packages</b>	

Copyright © 2002-2013 Avaya Inc. All rights reserved.

**Figure 22 – D-Channels Configuration Details**

Click on the **Basic Options (BSCOPT)** and click on the **Edit** button on the **Remote Capabilities** field as shown in **Figures 23**.

**AVAYA CS1000 Element Manager** Help | Logout

- UCMI 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

**- Basic options (BSCOPT)**

Action Device And Number (ADAN): DCH

D channel Card Type: DCIP

Designator: VoIP

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: 1800 Range: 0 - 3700

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**

- B channel Service messaging: ☐

**Change protocol timer value (TIMR)**

**Advanced options (ADVOPT)**

**Feature Packages**

Submit Refresh Delete Cancel

Copyright © 2002-2011 Avaya Inc. All rights reserved.

**Figure 23 – D-Channel Configuration Details**

The **Remote Capabilities Configuration** page appears as shown in **Figures 24**. Check on the **ND2** and the **MWI** checkboxes.

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

Managing: 10.10.97.96

Username: admin

Routes and Trunks » D-Channels » D-Channels 100 Property Configuration » Remote Capabilities Configuration

Remote Capabilities Configuration

Input Description	Input Value
Basic rate interface (BRI)	<input type="checkbox"/>
Call completion on busy using integer value (CCBI)	<input type="checkbox"/>
Call completion on busy using object identifier (CCBO)	<input type="checkbox"/>
Call completion on busy for QSIG and EuroISDN BRI (CCBS)	<input type="checkbox"/>
Call completion on no response using integer value (CCNI)	<input type="checkbox"/>
Call completion on no response using object identifier (CCNO)	<input type="checkbox"/>
Call completion to no reply for QSIG and EuroISDN BRI (CCNR)	<input type="checkbox"/>
Network call park (CPK)	<input type="checkbox"/>
Connected line identification presentation (COLP)	<input type="checkbox"/>
Call transfer integer (CTI)	<input type="checkbox"/>
Call transfer object (CTO)	<input type="checkbox"/>
Diversion info. is sent using integer value (DV1I)	<input type="checkbox"/>
Diversion info. is sent using object identifier (DV1O)	<input type="checkbox"/>
Rerouting requests processed using integer value (DV2I)	<input type="checkbox"/>
Rerouting requests processed using object identifier (DV2O)	<input type="checkbox"/>
Diversion info. sent. rerouting requests processed (DV3I)	<input type="checkbox"/>
EuroISDN - div. info sent. rerouting req. processed (DV3O)	<input type="checkbox"/>
Call transfer notification and invocation to EuroISDN (ECTO)	<input type="checkbox"/>
Malicious call identification (MCID)	<input type="checkbox"/>
MCDN QSIG conversion (MQC)	<input type="checkbox"/>
Remote D-channel is on a MSDL card (MSL)	<input type="checkbox"/>
Message waiting interworking with DMS-100 (MWI)	<input checked="" type="checkbox"/>
Network access data (NAC)	<input type="checkbox"/>
Network call trace supported (NCT)	<input type="checkbox"/>
Network name display method 1 (ND1)	<input type="checkbox"/>
Network name display method 2 (ND2)	<input checked="" type="checkbox"/>
Network name display method 3 (ND3)	<input type="checkbox"/>
Name display - integer ID coding (NDI)	<input type="checkbox"/>
Name display - object ID coding (NDO)	<input type="checkbox"/>

Copyright © 2002-2011 Avaya Inc. All rights reserved.

**Figure 24 – Remote Capabilities Configuration Details**

Click on the **Return – Remote Capabilities** button (not shown).

Click on the **Submit** button (not shown).

#### 5.5.4. Administer Virtual Super-Loop

Select **System** → **Core Equipment** → **Superloops** from the left pane to display the **Superloops** screen. If the Superloop does not exist, please click the **Add** button to create a new one as shown in **Figure 25**. In this example, Superloop 4, 96, 100, and 124 have been added and are being used.

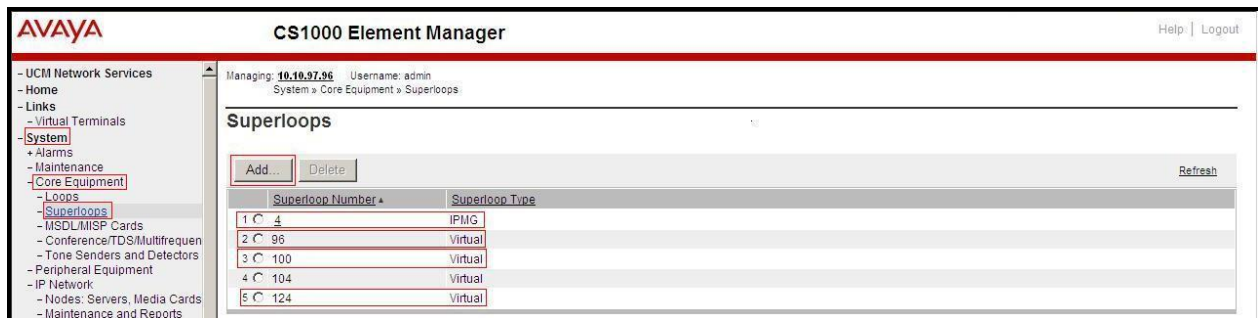


Figure 25 – Administer Virtual Super-Loop Page

#### 5.5.5. Administer Virtual SIP Routes

Select **Routes and Trunks** → **Routes and Trunks** (not shown) 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 26**.

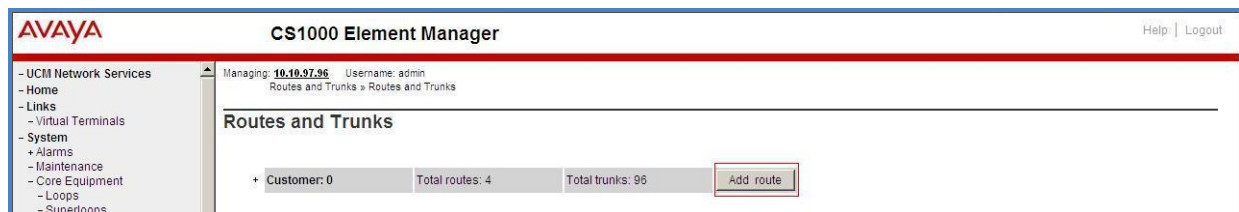


Figure 26 – Add route

The **Customer 0, New Route Configuration** screen is displayed next (not shown). The **Basic Configuration** section is displayed to put the following values for the specific fields, and retain the default values for the remaining fields. The screenshot of Basic Configuration section of existing route 100 is displayed to edit as shown in **Figures 27**.

- **Route number (ROUT):** Select an available route number (example: route **100**).
- **Designator field for trunk (DES):** A descriptive text (**100**).
- **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 (example: **8100**).
- Check the **The route is for a virtual trunk route (VTRK)** field, 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**). **Note:** The Zone value is filled out as 255, but after it is added, the screen is displayed with prefix 00.
- For the **Node ID of signaling server of this route (NODE)** field, enter the node number **3000** (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. Scrolling down to the bottom of the screen, enter the following values for the specified fields, and retain the default values for the remaining fields.
  - **Mode of operation (MODE):** Select Route uses ISDN Signalling Link (**ISLD**)
  - **D channel number (DCH):** Enter **100** (created in **Section 5.5.3**)
  - **Network calling name allowed (NCNA):** Check the field.
  - **Network call redirection (NCRD):** Check the field.
  - **Insert ESN access code (INAC):** Check the field.

**AVAYA CS1000 Element Manager**

Managing: 10.10.97.96 Username: admin  
Routes and Trunks » Routes and Trunks » Customer 0, Route 100 Property Configuration

**Customer 0, Route 100 Property Configuration**

**Basic Configuration**

Route data block (RDB) (TYPE): RDB  
Customer number (CUST): 00  
Route number (ROUT): 100  
Designator field for trunk (DES): 100  
Trunk type (TKTP): TIE  
Incoming and outgoing trunk (ICOG): Incoming and Outgoing (IAO)  
Access code for the trunk route (ACOD): 8100  
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): 3000 (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 DT12 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): ☒

Copyright © 2002-2011 Avaya Inc. All rights reserved.

**Figure 27 – Route Configuration Details**

Click on **Basic Route Options**, check the **North American toll scheme (NATL)** and **Incoming DID digit conversion on this route (IDC)** checkboxes. Enter **1** for both **Day IDC tree number** and **Night IDC tree number** as shown in **Figure 28**. Click on the **Submit** button.

**AVAYA CS1000 Element Manager**

Help | Logout

- UCM Network Services  
- Home  
- Links  
- Virtual Terminals  
- System  
+ Alarms  
- Maintenance  
- Core Equipment  
- Loops  
- Superloops  
- MSDLMISP Cards  
- Conference/TDS/Multifrequen  
- Tone Senders and Detectors  
- Peripheral Equipment  
- IP Network  
- Nodes: Servers, Media Cards  
- Maintenance and Reports  
- Media Gateways  
- Zones  
- Host and Route Tables  
- Network: Address Translation  
- 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

- Mobile extension timer (MBXT): 0 (0 - 8000 milliseconds)  
Calling number dialing plan (CNDP): Unknown (UKWN)

**Basic Route Options**

Attendant announcement (ATAN): No Attendant Announcement (NO)  
Billing number required (BLN): ☐  
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 ACC 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  
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

Copyright © 2002-2011 Avaya Inc. All rights reserved.

**Figure 28 – Route Configuration Details**

### 5.5.6. Administer Virtual Trunks

Select **Routes and Trunks** → **Route and Trunks** (not shown). The Route list is now updated with the newly added routes. In the example, the Route 100 was being added. Click on the **Add trunk** button as shown in **Figure 29**.

**AVAYA CS1000 Element Manager**

Help | Logout

Managing: 10.10.97.96 Username: admin  
Routes and Trunks → Routes and Trunks

**Routes and Trunks**

Customer	Total routes	Total trunks	
Customer: 0	Total routes: 4	Total trunks: 96	Add route
+ Route: 11	Type: TIE	Description: SIPL	Edit Add trunk
+ Route: 100	Type: TIE	Description: 100	Edit Add trunk

**Figure 29 – Routes and Trunks Page**

The **Customer 0, Route 100, Trunk 1 Property Configuration** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields. The Media Security (sRTP) needs to be disabled at the trunk level by editing the **Class of Service (CLS)** at the bottom of the basic trunk configuration page. Click on the **Edit** button as shown in **Figure 30**.

**Note:** 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:** IP Trunk (**IPTI**)
- **Terminal Number:** Available terminal number (Superloop 100 created in **Section 5.5.4**)
- **Designator field for trunk:** A descriptive text
- **Extended Trunk:** Virtual trunk (**VTRK**)
- **Member number:** Current route number and starting member
- **Card Density:** 8D
- **Start arrangement Incoming:** Immediate (**IMM**)
- **Start arrangement Outgoing:** Immediate (**IMM**)
- **Trunk group access restriction:** Desired trunk group access restriction level
- **Channel ID for this trunk:** An available starting channel ID

**Figure 30 – New Trunk Configuration Details**



For **Media Security**, select **Media Security Never (MSNV)**. Enter the values for the specified fields as shown in **Figure 31**. Scroll down to the bottom of the screen and click **Return Class of Service** and click on the **Save** button (shown in **Figure 30**).

**AVAYA CS1000 Element Manager**

Help | Logout

**- Class of Service**

Input Description	Input Value
- ACD Priority:	ACD Priority not required (APN)
- Analog Semi-Permanent Connections:	Analog Semi-Permanent Connections Denied (SPCD)
- ARF Supervised COT:	
- Barring:	
- Battery Supervised COT:	
- Busy Tone Supervised COT:	
- Calling Line Identification:	
- Calling party:	Calling party Denied (CND)
- Central Office Ringback:	
- Centrex Switchhook Flash:	Centrex Switchhook Flash Denied (THFD)
- Dial Pulse:	Digitone (DTN)
- DTR PAD value:	
- Echo Canceling:	Echo Canceling Denied (ECD)
- Hong Kong DTI:	
- Loop Break Supervised COT:	
- Make-break ratio for dial pulse:	10 pulses per second (P10)
- Manual Incoming:	Manual Incoming Denied (MID)
- Media Security:	Media Security Never (MSNV)
- Network Hook Flash Over M911P:	
- Polarity:	
- Priority:	Low Priority (LPR)
- Restriction level:	Unrestricted (UNR)
- Reversed Ear Piece:	Reversed Ear Piece denied (XREP)
- Short or long line:	
- Transmission Class of Service:	Non-Transmission Compensated (NTC)
- Warning Tone:	Warning Tone Allowed (WTA)
- Reversed Ear Piece:	Reversed Ear Piece denied (XREP)
- ARF Supervised COT:	

Return Class of Service Cancel

Copyright © 2002-2011 Avaya Inc. All rights reserved.

**Figure 31 – Class of Service Configuration Details Page**

## 5.5.7. Administer Calling Line Identification Entries

Select **Customers** → **00** → **ISDN and ESN Networking** on the left pane. Click on **Calling Line Identification Entries** as shown in Figure 32.

AVAYA CS1000 Element Manager

Managing: 10.10.97.96 Username: admin  
Customers > Customer 00 > Customer Details > ISDN and ESN Networking

### ISDN and ESN Networking

**General Properties**

Flexible trunk to trunk connection option:

Flexible orbiting prevention timer:

Country code:  (0 - 9999)

Code for processing the called number

National access code:

International access code:

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

**Calling Line Identification**

Information for incoming/outgoing calls:

Size:  (0 - 4000)

Country code:  (0 - 9999)

Code displayed as part of calling number

Calling Line Identification Entries

Save Cancel

Figure 32 – ISDN and ESN Networking

Click on **Add** as shown in Figure 33.

AVAYA CS1000 Element Manager

Managing: 10.10.97.96 Username: admin  
Customers > Customer 00 > Customer Details > ISDN and ESN Networking > Calling Line Identification Entries

### Calling Line Identification Entries

Search for CLID

Start range:

End range:

\*End range\* should not exceed the CLID size specified

Search

Calling Line Identification Entries

Add Delete Refresh

Figure 33 – Calling Line Identification Entries

The add entry **0** screen is displayed to put the following values for the specified fields and retain the default values for the remaining fields. The Edit Calling Line Identification of existing entry **0** is displayed as shown in **Figure 34**.

- **National Code:** leave it blank.
- **Local Code:** input prefix digits assigned by Windstream, in this case it is 6 digits – **678593**. This **Local Code** will be used for call display purpose for Call Type = Unknown.
- **Home Location Code:** input prefix digits assigned by Windstream, in this case it is 6 digits - **678593**. This **Home Location Code** will be used for call display purpose for Call Type = National (NPA).
- **Local Steering Code:** input prefix digits assigned by Windstream, in this case it is 6 digits - **678593**. This **Local Steering Code** will be used for call display purpose for Call Type = Local Subscriber (NXX).
- **Use DN as DID:** YES.
- **Calling Party Name Display:** Uncheck **Roman characters**.

Click on the **Save** button as shown in **Figure 34**

**AVAYA CS1000 Element Manager**

Managing: 10.10.97.96 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: 678593 (1-12 digits)  
Code for home local number or listed DN

Home Location Code: 678593 (1-7 digits)

Local Steering Code: 678593 (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:  12

Display Format: First name, Last name

**Save Cancel**

**Figure 34 – Edit Calling Line Identification 0**

### 5.5.8. Enable External Trunk to Trunk Transfer

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

Log in to Call Server Overlay CLI (please refer to **Section 5.1.2** for more details).

Allow External Trunk to Trunk Transfer for Customer Data Block by using **LD 15**.

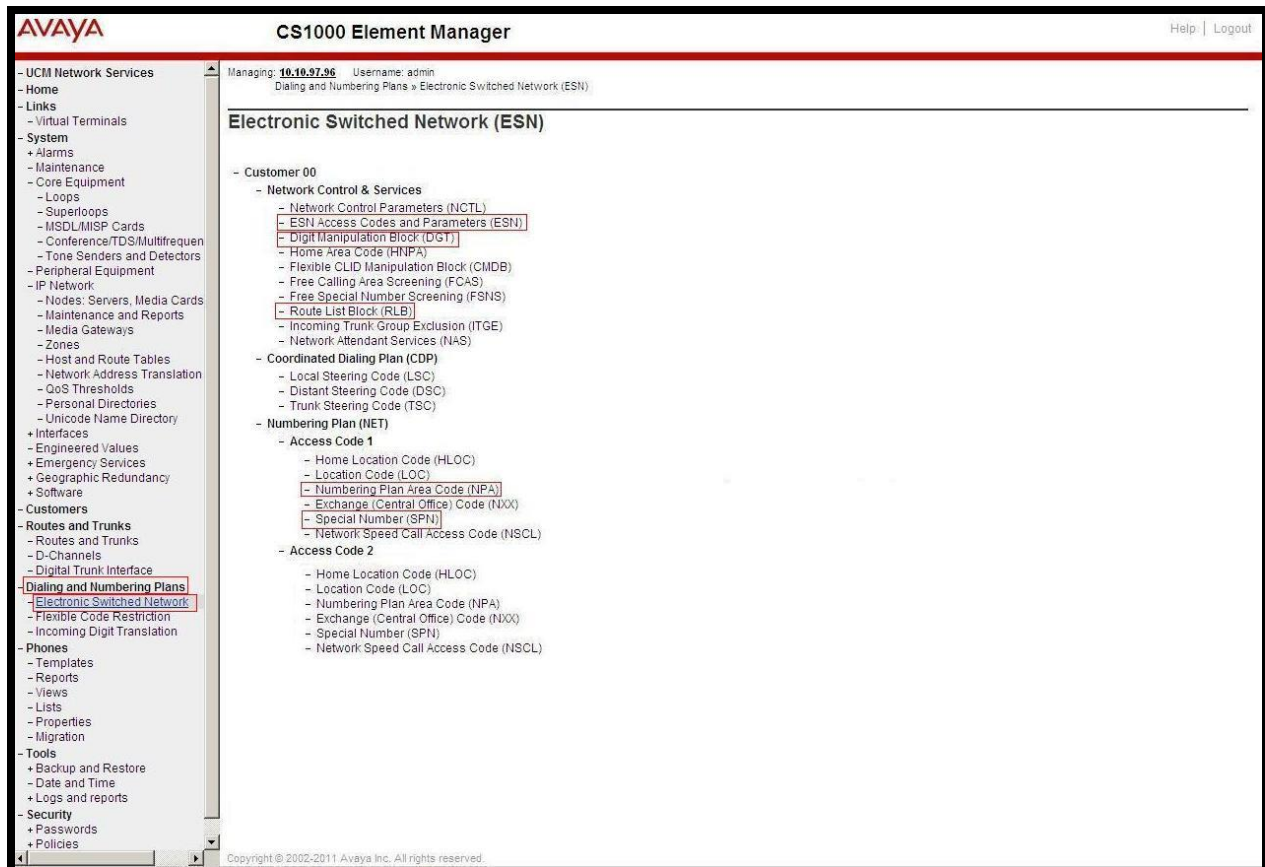
```
>ld 15
CDB000
MEM AVAIL: (U/P): 33600126   USED U P: 8345621 954062   TOT: 45579868
DISK SPACE NEEDED: 1722 KBYTES
REQ: chg
TYPE: net

TYPE NET_DATA
CUST 0
OPT
...
TRNX YES (←Enable transfer feature)
EXTT YES (← Enable external trunk to trunk Transfer )
...
```

## 5.6. Administer Dialing Plans

### 5.6.1. Define ESN Access Codes and Parameters (ESN)

Select **Dialing and Numbering Plans** → **Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen as shown in **Figure 35**.



**Figure 35 –ESN Configuration Details**

Select **ESN Access Codes and Basic Parameters** to define **NARS/BARS Access Code 1** as shown in **Figure 36**.

Click the **Submit** button (not shown).

**Figure 36 – ESN Access Codes and Basic Parameters**

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

Log in to Call Server CLI (please refer to **Section 5.1.2** for more details), change Customer Net Data block by using **LD 15**.

```
>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   → (Set NPA, SPN not to associate to ESN Access Code 2)
FNP
CLID
...
```

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 ----- > (NPA, SPN are associated to ESN Access Code 1)
AC2
FNP YES
...
```

### 5.6.3. Digit Manipulation Block Index (DMI)

The following steps show how to add DMI for the outbound call. There is an index, which was added to the Digit Manipulation Block List (14).

Select **Dialing and Numbering Plans** → **Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen as shown in **Figure 35**. Select **Digit Manipulation Block (DGT)**. The **Digit Manipulation Block List** is displayed as shown in **Figure 37**. In the **Please choose the** field, select an available **Digit Manipulation Block Index** from the drop-down list, and click on the **to Add** button.



**Figure 37 – Add a DMI**



The DMI\_14 screen will open. In this testing, no leading digits are to be deleted, therefore, enter **0** for the **Number of leading digits to be deleted** field and select **NPA (Numbering Plan Area)** for the **Call Type to be used by the manipulated digits** and then click on **Submit** button as shown in **Figure 38**.

**Figure 38 – DMI\_14 Configuration Details**

#### 5.6.4. Route List Block (RLB) (RLB 14)

This session shows how to add a RLB associated with the DMI created in **Section 5.6.3**. Select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen as shown in **Figure 35**. Select **Route List Block (RLB)**.

Enter an available value in the textbox for the **Please enter a route list index** (in this case 14) and click on the **to Add** button as shown in **Figure 39**. The screen shown in **Figure 40** will open.

**Figure 39 – Add a Route List Block.**



Enter the following values for the specified fields, and retain the default values for the remaining fields (**Figure 40**). Scroll down to the bottom of the screen, and click on the **Submit** button (not shown).

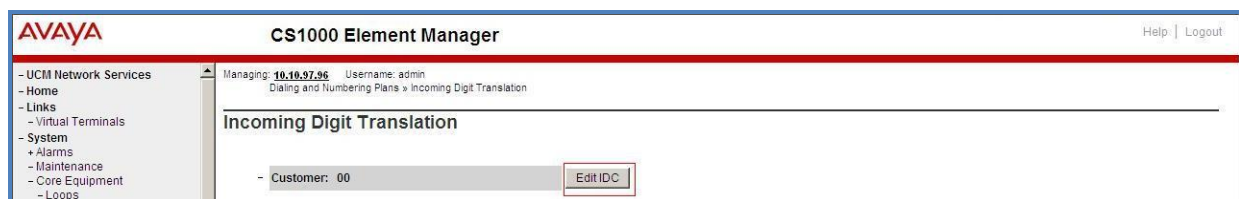
- **Digit Manipulation Index:** 14 (created in **Section 5.6.3**)
- **Incoming CLID Table:** 0 (created in **Section 5.5.7**)
- **Route number** 100 (created in **Section 5.5.5**)

**Figure 40 – RLB\_14 Route List Block Configuration Details**

### 5.6.5. Inbound Call – Incoming Digit Translation Configuration

This section describes the configuration steps required in order to receive calls from PSTN via the Windstream SIP Trunk Service.

Select **Dialing and Numbering Plans** → **Incoming Digit Translation** (not shown) from the left pane to display the **Incoming Digit Translation** screen. Click on the **Edit IDC** button as shown in **Figure 41**.



**Figure 41 – Incoming Digit Translation**

Click on the **New DCNO** to create the digit translation mechanism. In this example, **Digit Conversion Tree Number 1** has been created as shown in **Figure 42**.



**Figure 42 – Incoming Digit Conversion Property**

Detail configuration of the Digit Conversion Tree Configuration is shown in **Figure 43**. The **Incoming Digits** can be added to map to the Converted Digits which would be the associated Communication Server 1000 system phone DN. This **DCNO** has been assigned to route 100 as shown in **Figure 28**.

In the following configuration, the incoming call from PSTN with DID with prefix 678593 will be translated to the associated DN with 4 digits. For voicemail testing purposes, configure the DID number 6785931643 to translate to 1700.

Managing: 10.10.97.96 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	6785931641	1641	.	Roman characters
2	6785931642	1642	.	Roman characters
3	6785931643	1700	.	Roman characters

**Figure 43 – Digit Conversion Tree**

### 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, 1800, 411, 911 and so on.

Select **Dialing and Numbering Plans** → **Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen as show in **Figure 35**. Select **Special Number (SPN)**. Enter a SPN number and then click on **to Add** button. **Figure 44** shows all the special numbers used for this testing.

The screenshot displays the AVAYA CS1000 Element Manager web interface. The left-hand navigation pane lists various system components, with 'Dialing and Numbering Plans' and its sub-item 'Electronic Switched Network' highlighted. The main content area, titled 'Special Number List', shows a list of configured special numbers. At the top of this area, there is a form with the label 'Please enter a Special Number' followed by a text input field and a 'to Add' button. Below this, four special numbers are listed: 0, 1800, 411, and 911. Each entry includes an 'Edit' button and displays configuration details: 'Flexible length', 'Inhibit time-out handler' (set to NO), 'Type of call that is defined by the special number' (set to NONE), and 'Route list index' (set to 14).

Special Number	Flexible length	Inhibit time-out handler	Type of call that is defined by the special number	Route list index
0	14	NO	NONE	14
1800	11	NO	NONE	14
411	3	NO	NONE	14
911	3	NO	NONE	14

**Figure 44 – Add a SPN**

### 5.6.7. Outbound Call - Numbering Plan Area (NPA)

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

Select **Dialing and Numbering Plans** → **Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen (not shown). Select **Numbering Plan Area Code (NPA)** as shown in **Figure 35**. Enter the area code desired in the textbox and click on the **to Add** button. The 1214, 1281, 1303, 1613, 1720, 1832, 1972, and 613 area codes were used in this configuration as shown in **Figure 45**.

The screenshot displays the AVAYA CS1000 Element Manager web interface. The left-hand navigation pane lists various system components, with 'Dialing and Numbering Plans' and its sub-item 'Electronic Switched Network' highlighted. The main content area, titled 'Numbering Plan Area Code List', shows a list of configured area codes. At the top, there is a form to add a new area code with a text input field and a 'to Add' button. Below this, a table lists the following area codes: 1214, 1281, 1303, 1613, 1720, 1832, 1972, and 613. Each entry includes an 'Edit' button and associated configuration details such as 'Route List Index: 14' and 'Incoming Trunk group Exclusion Index: NONE'.

Numbering Plan Area Code	Route List Index	Incoming Trunk group Exclusion Index
1214	14	NONE
1281	14	NONE
1303	14	NONE
1613	14	NONE
1720	14	NONE
1832	14	NONE
1972	14	NONE
613	14	NONE

**Figure 45 – Numbering Plan Area Code List**

## 5.7. Administer a Phone

This section describes the creation of Communication Server 1000 clients used in this configuration.

### 5.7.1. Phone creation

Refer to **Section 5.5.4** to create a Virtual Superloop - **96** used for IP phones. Refer to **Section 5.4.1** to create a bandwidth zone - **10** for IP phones. Log in to the Call Server Command Line Interface (please refer to **Section 5.1.2** for more detail). Create an IP phone by using LD 11 as shown below:

```
REQ: prt
TYPE: 2002p2
TN 96 0 0 2
DATE
PAGE
DES
MODEL_NAME
EMULATED
DES 2002P2 < --- Describe information for IP Phone
TN 96 0 00 02 VIRTUAL < --- Set Terminal Number for IP Phone
TYPE 2002P2
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 00010 < --- Set bandwidth zone for IP phone
CUR_ZONE 00010
MRT
ERL 12345
ECL 0
FDN
TGAR 0
LDN NO
NCOS 7
SGRP 0
RNPG 0
SCI 0
SSU
LNRS 16
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS UNR FBA WTA LPR MTD FNA HTA TDD CRPD
    MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
    POD SLKD CCSD SWD LNA CNDA
    CFTD SFA MRD DDV CNIA CDCA MSID DAPA BFED RCBF
    ICDD CDMD LLCN MCTD CLBD AUTU
    GPUD DPUD DNDD CFXA 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 CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD
MSNV FRA PKCH MWTD DVLD CROD ELCD
CPND_LANG ENG
HUNT
PLEV 02
PUID
UPWD
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 SCR 1641 0   MARP < --- Set the position of DN 1641 to display on key 0 of the phone
    CPND
        CPND_LANG ROMAN
            NAME Winds1 < --- Set name to display
            XPLN 13
            DISPLAY_FMT FIRST, LAST
    01
<Text removed for brevity>

```

### 5.7.2. Enable Privacy for the Phone

This section shows how to enable Privacy for a phone by changing its class of service (CLS) and this feature cannot be enabled or disabled from the phone. By modifying the configuration of the phone created in **Section 5.7.1**, the display of the outbound call will be changed appropriately.

To hide the display number, set **CLS** (Class of Service) to **DDGD**. Communication Server 1000 will include “Privacy:id” in the SIP message header before sending it to Windstream.

```

>ld 11
REQ: chg
TYPE: 2002p2
TN 96 0 0 2
ECHG yes
ITEM CLS DDGD
...

```



To allow the display number, set **CLS** to **DDGA**. Communication Server 1000 will not send the Privacy header to Windstream.

```
>ld 11
REQ: chg
TYPE: 2002p2
TN 96 0 0 2
ECHG yes
ITEM CLS DDGA
...
```

### 5.7.3. Enable Call Forward for Phone

This section shows how to configure the Call Forward feature at the system and phone level.

Select **Customer → 00 → Call Redirection**. The Call Redirection page is shown in **Figure 46**.

- **Total redirection count limit: 0** (unlimited)
- **Call Forward: Originating**
- **Number of normal ring cycle for CFNA: 3**
- Click **Save** to save the configuration.

**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

Call redirection by day: ☐

Days for day option 0:

Days for day option 1:

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 46 – Call Redirection**

To enable **Call Forward All Call (CFAC)** feature for a phone over SIP trunk, use **LD 11**. Change its **CLS** to **CFXA**, and **SFA**, then program the forward number on the phone set. The following is the configuration of a phone that has CFAC enabled with forwarding number 613034581222.

```
ld 11
REQ: chg
TYPE: 2007
TN 96 0 0 4

ECHG yes
ITEM CLS CFXA SFA
ITEM key 19 CFW 16 613034581222
```

To enable **Call Forward Busy (CFB)** feature for phone over SIP trunk, use **LD 11**. Change its **CLS** to **FBA**, **HTA**, and **SFA**, then program the forward number as **HUNT** and **FDN**. The following is the configuration of a phone has **CFB** enabled with forwarding number 613034581222.

```
ld 11
REQ: chg
TYPE: 2007
TN 96 0 0 4
ECHG yes
ITEM CLS FBA HTA SFA
ITEM HUNT 613034581222
ITEM FDN 613034581222
```

To enable **Call Forward No Answer (CFNA)** feature for a phone over SIP trunk, use **LD 11**. Change its **CLS** to **FNA**, and **SFA**, then program the forward number as **HUNT** and **FDN**. The following is the configuration of a phone that has CFNA enabled with forwarding number 613034581222.

```
ld 11
REQ: chg
TYPE: 2007
TN 96 0 0 4
ECHG yes
ITEM CLS FNA SFA
ITEM HUNT 613034581222
ITEM FDN 613034581222
```

#### 5.7.4. Enable Call Waiting for Phone

This section shows how to configure the Call Waiting feature at the phone level.

Log in to the Call Server CLI (please refer to **Section 5.1.2** for more details), configure Call Waiting feature for phone by using **LD 11** to change **CLS** to **HTD**, and **SWA** and adding a **CWT** key.

```
ld 11
REQ: chg
TYPE: 2002p2
TN 96 0 0 2
ECHG yes
ITEM CLS HTD SWA
ITEM key 2 CWT
```

## 6. Configure Session Border Controller for Enterprise

This section describes the configuration of the Avaya SBCE necessary for interoperability with the Communication Server 1000 and Windstream SIP Trunk Service.

Avaya elements reside on the Private side and the Windstream SIP Trunk Service reside on the Public side of the network, as illustrated in **Figure 1**.

**Note:** The following section assumes that Avaya SBCE has been installed and that network connectivity exists between the systems. For more information on Avaya SBCE, see **Section 10** of these Application Notes.

### 6.1. Log into the Avaya SBCE

Access the web interface by typing “<https://x.x.x.x/sbc/>” (where x.x.x.x is the management IP of the Avaya SBCE).

Enter the **Username** and **Password**.

**AVAYA**

**Session Border Controller  
for Enterprise**

**Log In**

Username:

Password:

This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use or modifications of this system is strictly prohibited. Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

All users must comply with all corporate instructions regarding the protection of information assets.

© 2011 - 2013 Avaya Inc. All rights reserved.

**Figure 47 - Avaya SBCE Login**

## 6.2. Global Profiles

When selected, Global Profiles allows for configuration of parameters across all Avaya SBCE appliances.

### 6.2.1. Configure Server Interworking - Avaya site

Server Interworking allows one to configure and manage various SIP call server-specific capabilities such as call hold, 180 handling, etc.

From the menu on the left-hand side, select **Global Profiles** → **Server Interworking** → **Add**

- Enter Profile name: **CS1K76**
- On the **General** tab, set **180 Handling** as **No SDP**. Other options can be left at default.

On the **Timers**, **URI Manipulation**, **Header Manipulation** and **Advanced** tabs: All options can be left at default. Click **Finish** (not shown).

The following screen is shown that Communication Server 1000 server interworking (named: **CS1K76**) was added.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes links for Alarms, Incidents, Statistics, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header shows the product name and the Avaya logo. On the left, a sidebar menu lists various configuration areas, with 'Server Interworking' highlighted under 'Global Profiles'. The main content area is titled 'Interworking Profiles: CS1K76' and features an 'Add' button. Below this, a list of existing profiles is shown, with 'CS1K76' selected. The configuration tabs include General, Timers, URI Manipulation, Header Manipulation, and Advanced. The 'General' tab is active, showing a table of parameters and their values. The '180 Handling' parameter is set to 'No SDP'. The 'Privacy' section is also visible at the bottom of the configuration table.

General	
Hold Support	NONE
180 Handling	No SDP
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	No
URI Group	None
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
Re-Invite Handling	No
T.38 Support	No
URI Scheme	SIP
Via Header Format	RFC3261

Privacy	
Privacy Enabled	No
User Name	
P-Asserted-Identity	No
P-Preferred-Identity	No
Privacy Header	

**Figure 48 - Server Interworking – Avaya site**

## 6.2.2. Configure Server Interworking – Windstream site

From the menu on the left-hand side, select **Global Profiles** → **Server Interworking** → **Add**

- Enter Profile name: **SP3**
- All options on the **General** tab can be left at default.

On the **Timers**, **URI Manipulation**, **Header Manipulation** and **Advanced** tabs: All options can be left at default. Click **Finish** (not shown).

The following screen is shown that Windstream server interworking (named: **SP3**) was added.

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes links for Alarms, Incidents, Statistics, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header shows the title "Session Border Controller for Enterprise" and the Avaya logo. On the left, a sidebar menu lists various configuration areas, with "Global Profiles" expanded and "Server Interworking" highlighted. The main content area is titled "Interworking Profiles: SP3" and features an "Add" button. Below this, a list of existing profiles is shown, including "cs2100", "avaya-ru", "OCS-Edge-Server", "cisco-ccm", "cups", "OCS-FrontEnd-Server", "CS1K76", and "SP3". The "SP3" profile is selected, and its configuration is displayed in a tabbed interface. The "General" tab is active, showing a table of settings. The "Privacy" tab is also visible, showing settings for "Privacy Enabled", "User Name", "P-Asserted-Identity", "P-Preferred-Identity", and "Privacy Header".

General	
Hold Support	NONE
180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	No
URI Group	None
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
Re-Invite Handling	No
T.38 Support	No
URI Scheme	SIP
Via Header Format	RFC3261

Privacy	
Privacy Enabled	No
User Name	
P-Asserted-Identity	No
P-Preferred-Identity	No
Privacy Header	

Figure 49 - Server Interworking – Windstream site

### 6.2.3. Configure URI Groups

The URI Group feature allows administrator to create any number of logical URI groups that are comprised of individual SIP subscribers located in that particular domain or group.

The following URI Group configuration is used for this specific testing in DevConnect Lab environment. The URI-Group named **SP3** was used to match the “From” and “To” headers in a SIP call dialog received from both Enterprise and Windstream SIP Trunk Service. If there is a match, the Avaya SBCE will apply the appropriate Routing profiles (see **Section 6.2.4, 6.2.5**), Server Flows (see **Section 6.4.4**), and Session Flow (see **section 6.4.5**) to route incoming and outgoing calls to the right destinations. In production environment, there is not a requirement to define this URI.

From the menu on the left-hand side, select **Global Profiles → URI Groups**. Select **Add**.

- Enter Group Name: **SP3**.
- Edit the URI Type: **Regular Expression** (not shown).
- **Add URI**: **.\*10\10\97\178** (Communication Server 1000 Node IP address), **.\*10\10\98\111** (Avaya SBCE public interface IP address), **.\*10\10\98\13** (Avaya SBCE internal interface IP address), **.\*192\168\64\229** (Windstream SIP Signaling server IP address), **.\*bvwddev7\com** (Enterprise domain), **.\*mcleodusa** (Windstream domain), **.\*nonymous** (Anonymous URI).
- Click **Finish** (not shown).

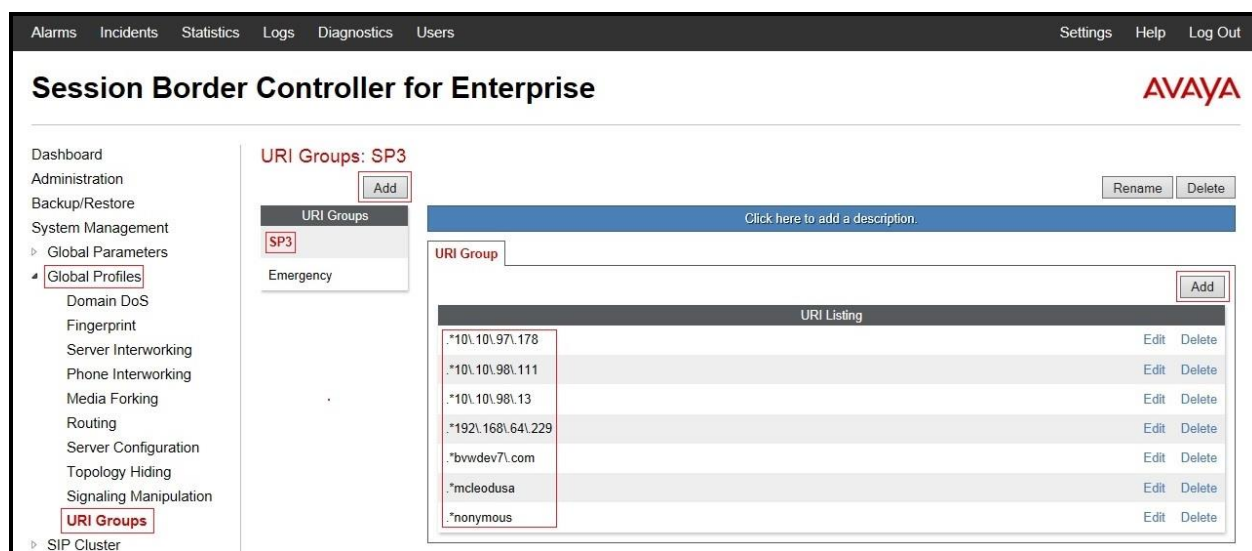


Figure 50 - URI Group

### 6.2.4. Configure Routing – Avaya site

Routing profiles define a specific set of packet routing criteria that are used in conjunction with other types of domain policies to identify a particular call flow and thereby ascertain which security features will be applied to those packets. Parameters defined by Routing Profiles include

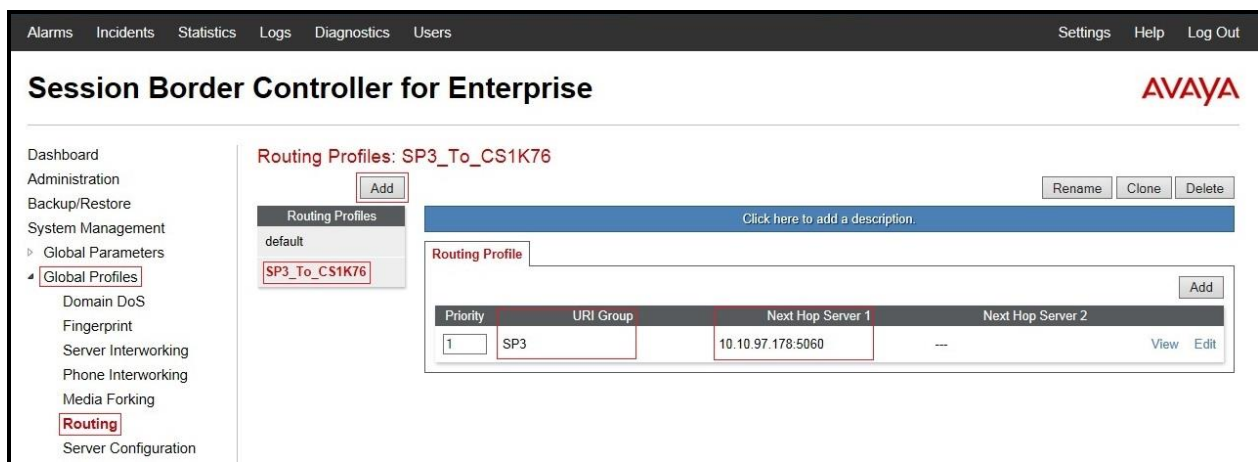


packet transport settings, name server addresses and resolution methods, next hop routing information, and packet transport types.

From the menu on the left-hand side, select **Global Profiles → Routing → Add**

Enter Profile Name: **SP3\_To\_CS1K76**

- **URI Group: SP3**
- **Next Hop Server 1: 10.10.97.178:5060** (Communication Server 1000 Node IP address)
- Check **Routing Priority based on Next Hop Server** (not shown)
- **Outgoing Transport: UDP** (not shown)
- Click **Finish** (not shown).



**Figure 51 - Routing to Avaya**

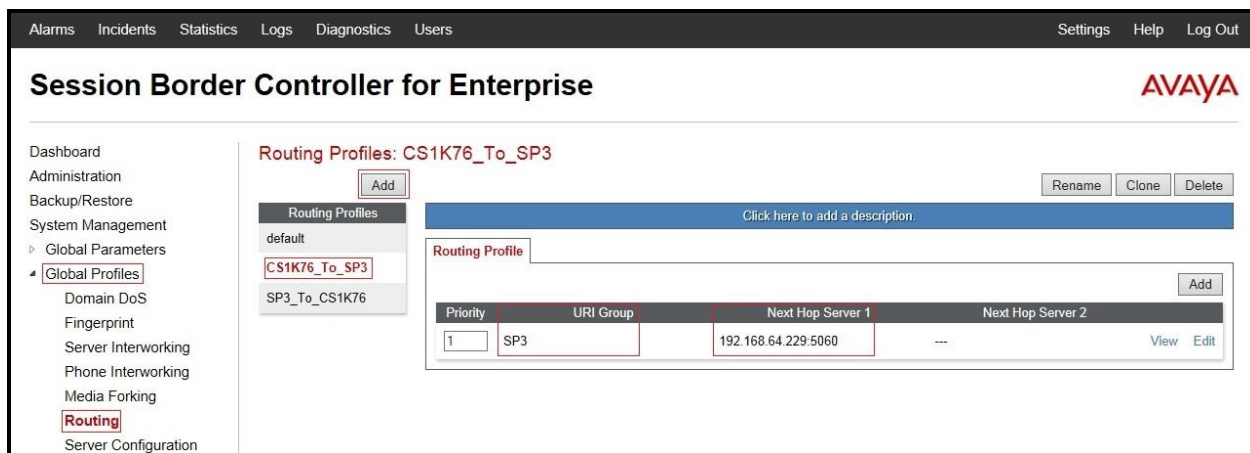
### 6.2.5. Configure Routing – Windstream site

The Routing Profile allows one to manage parameters related to routing SIP signaling messages.

From the menu on the left-hand side, select **Global Profiles → Routing → Add**

Enter Profile Name: **CS1K76\_To\_SP3**

- **URI Group: SP3**
- **Next Hop Server 1: 192.168.64.229:5060** (Windstream SIP Signaling server IP address)
- Check **Routing Priority based on Next Hop Server** (not shown)
- **Outgoing Transport: UDP** (not shown)
- Click **Finish** (not shown).



**Figure 52 - Routing to Windstream**

### 6.2.6. Configure Signaling Manipulation

The Avaya's SIP signaling header manipulation feature is used for the Avaya SBCE product. This feature adds the ability to add, change and delete any of the headers and other information in a SIP message.

- Select **Global Profiles** from the menu on the left-hand side
- Select the **Signaling Manipulation**
- Select **Add**. Enter script Title: **SP3**
  - Edit the script to remove unexpected Diversion Header SIP message from incoming calls.
  - Edit the script to replace History Info by Diversion Header for call forward off-net.
  - Edit the script to replace MIME by SDP.
  - Edit the script to remove unwanted SIP headers.
  - Click **Save** (not shown).

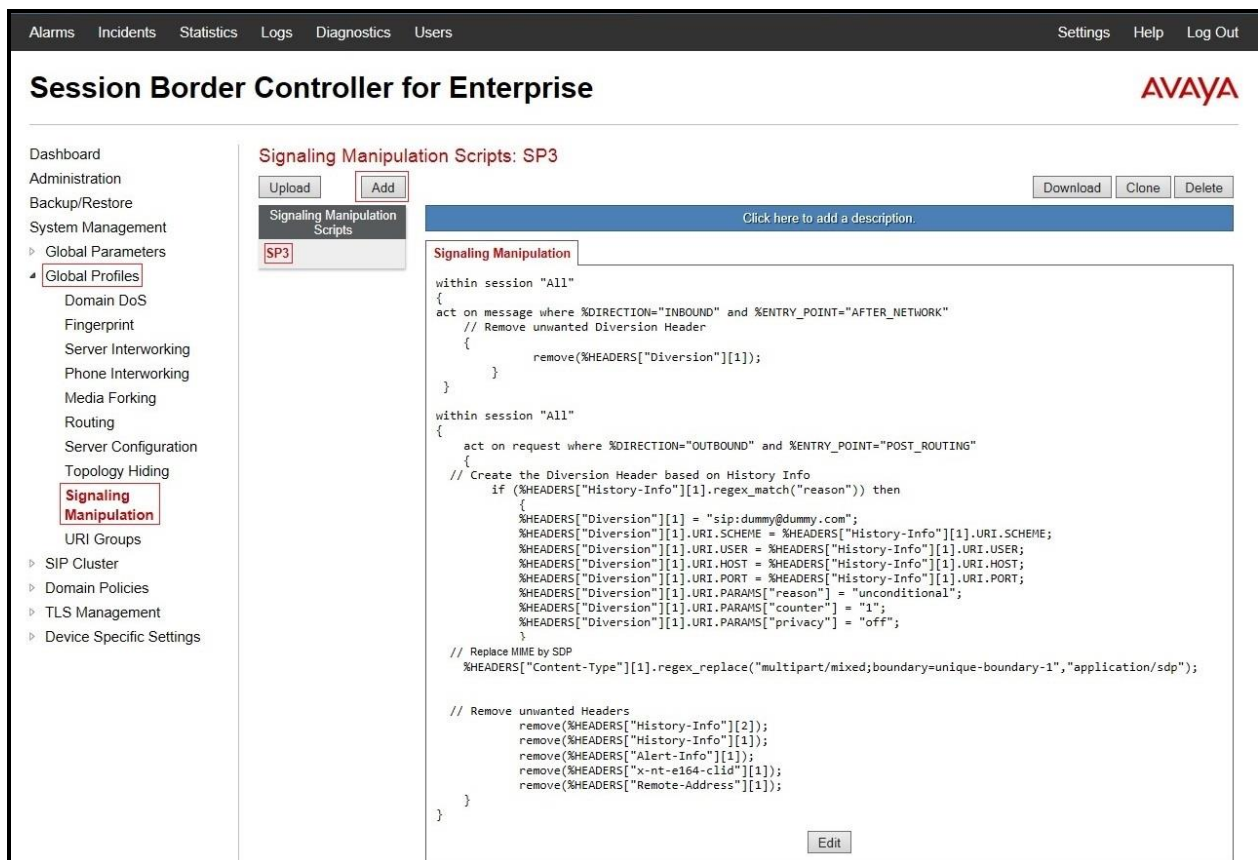


Figure 53 – Signaling Manipulation Windstream

### 6.2.7. Configure Server – Communication Server 1000

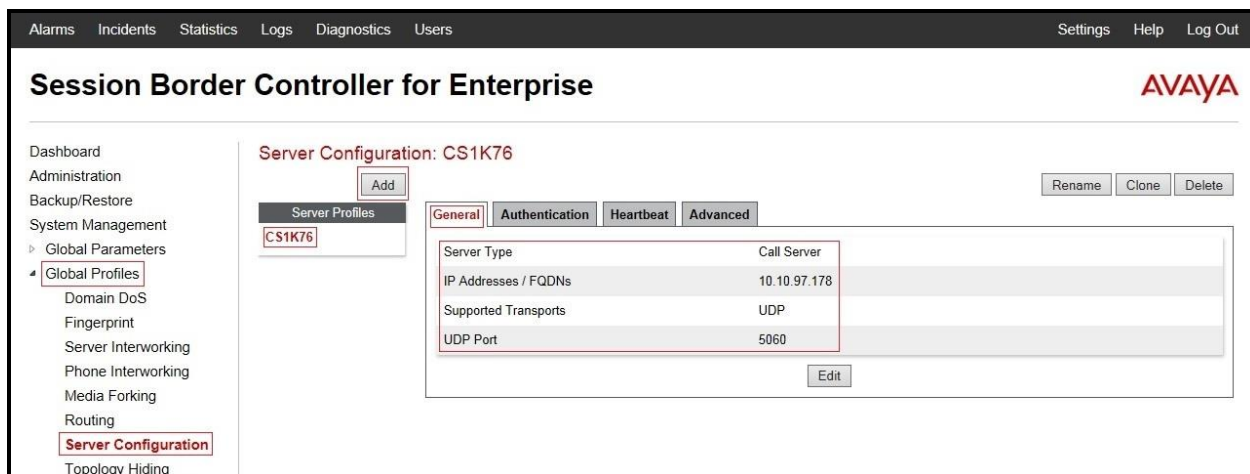
The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs allow one to configure and manage various SIP call server-specific parameters such as UDP port assignment, IP Server type, heartbeat signaling parameters and some advanced options.

From the menu on the left-hand side, select **Global Profiles → Server Configuration → Add**.

Enter profile name: **CS1K76**

On **General** tab, enter the following:

- **Server Type:** Select **Call Server**
- **IP Address/FQDNs:** **10.10.97.178** (Communication Server 1000 Node IP Address)
- **Supported Transports:** **UDP**
- **UDP Port:** **5060**

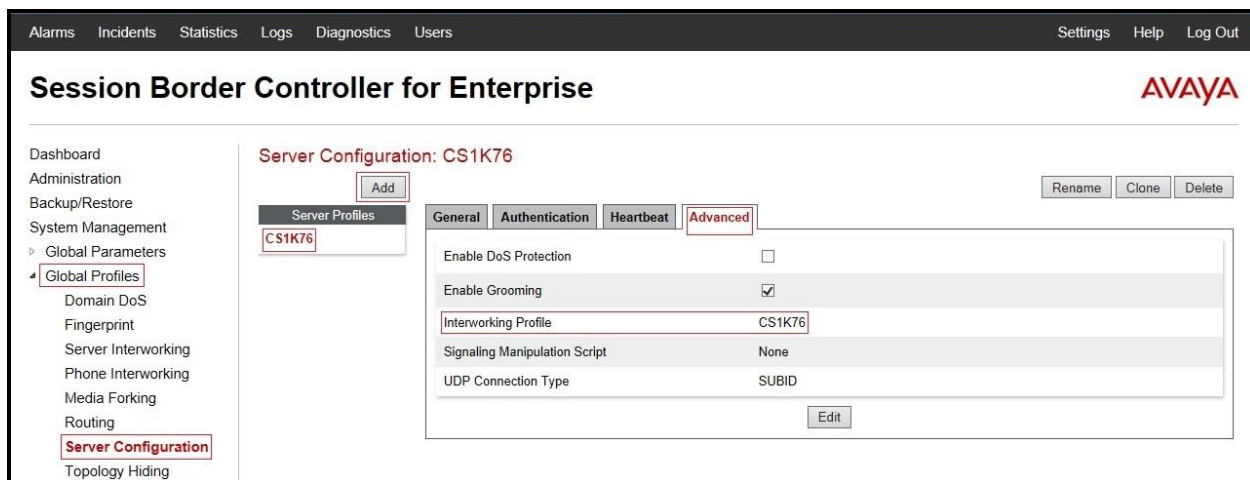


**Figure 54 – Communication Server 1000 General Server Configuration**

On the **Advanced** tab:

- Select **CS1K76** for **Interworking Profile**

Click **Finish** (not shown).



**Figure 55 – Communication Server 1000 Advanced Server Configuration**

### 6.2.8. Configure Server – Windstream

From the menu on the left-hand side, select **Global Profiles** → **Server Configuration** → **Add**.

Enter profile name: **SP3**

On **General** tab, enter the following:

- **Server Type:** Select **Trunk Server**
- **IP Address:** **192.168.64.229** (Windstream Signaling server IP Address)

- **Supported Transports: UDP**
- **UDP Port: 5060**

The screenshot shows the 'Session Border Controller for Enterprise' interface. The left sidebar contains a navigation menu with 'Server Configuration' highlighted. The main content area is titled 'Server Configuration: SP3'. It features a 'Server Profiles' list on the left with 'SP3' selected. The 'General' tab is active, displaying a table of configuration details:

Field	Value
Server Type	Trunk Server
IP Addresses / FQDNs	192.168.64.229
Supported Transports	UDP
UDP Port	5060

Buttons for 'Add', 'Rename', 'Clone', 'Delete', and 'Edit' are visible.

**Figure 56 - Windstream General Server Configuration**

On the **Advanced** tab, enter the following:

- **Interworking Profile:** select **SP3**
- **Signaling Manipulation Script:** select **SP3**

Click **Finish** (not shown).

The screenshot shows the 'Advanced' tab of the 'Server Configuration: SP3' page. The configuration details are as follows:

Field	Value
Enable DoS Protection	<input type="checkbox"/>
Enable Grooming	<input checked="" type="checkbox"/>
Interworking Profile	SP3
Signaling Manipulation Script	SP3
UDP Connection Type	SUBID

Buttons for 'Add', 'Rename', 'Clone', 'Delete', and 'Edit' are visible.

**Figure 57 - Windstream Advanced Server Configuration**

### 6.2.9. Configure Topology Hiding – Avaya site

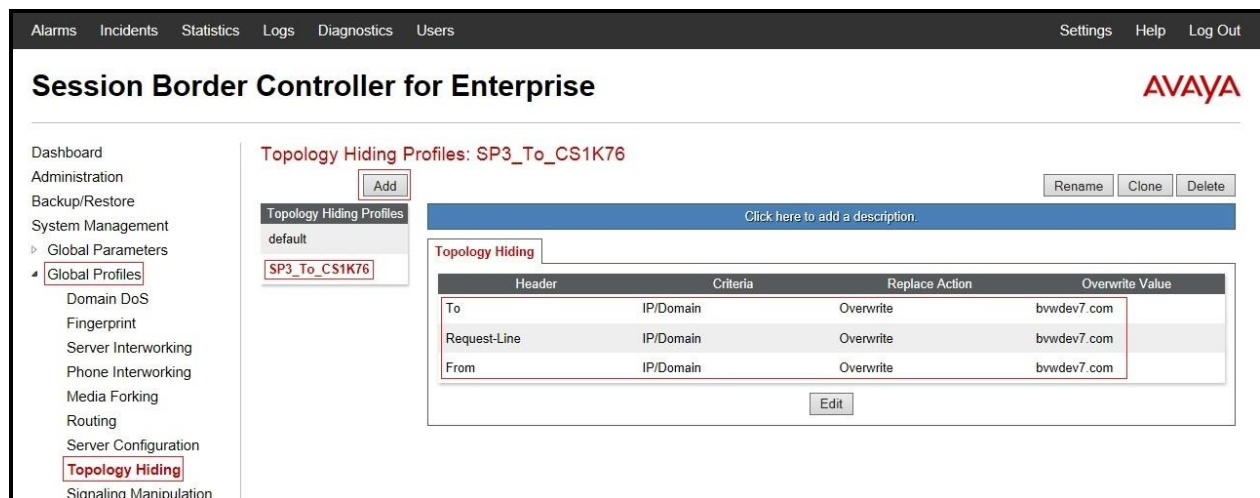
The Topology Hiding screen allows one to manage how various source, destination and routing information in SIP and SDP message headers are substituted or changed to maintain the integrity of the network. It hides the topology of the enterprise network from external networks

From the menu on the left-hand side, select **Global Profiles** → **Topology Hiding**.

Select **Add**, enter Profile Name: **SP3\_To\_CS1K76**.

- For the Header **To**,
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select: **Overwrite**  
In the **Overwrite Value** column: **bwdev7.com**
- For the Header **Request-Line**,
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select: **Overwrite**
  - In the **Overwrite Value** column: **bwdev7.com**
- For the Header **From**,
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select: **Overwrite**  
In the **Overwrite Value** column: **bwdev7.com**

Click **Finish** (not shown).



**Figure 58 - Topology Hiding Communication Server 1000**



### 6.2.10. Configure Topology Hiding – Windstream site

From the menu on the left-hand side, select **Global Profiles** → **Topology Hiding**.

Select **Add Profile**, enter Profile Name: **CS1K76\_To\_SP3**.

- For the Header **To**,
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select: **Overwrite**
  - In the **Overwrite Value** column: **192.168.64.229**
- For the Header **Request-Line**,
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select: **Overwrite**
  - In the **Overwrite Value** column: **192.168.64.229**
- For the Header **From**,
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select: **Overwrite**
  - In the **Overwrite Value** column: **10.10.98.111**

Click **Finish** (not shown).

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The left-hand navigation menu includes options like Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, Domain DoS, Fingerprint, Server Interworking, Phone Interworking, Media Forking, Routing, Server Configuration, Topology Hiding, and Signaling Manipulation. The 'Topology Hiding' option is highlighted. The main content area displays 'Topology Hiding Profiles: CS1K76\_To\_SP3'. A table lists the configured profiles: 'default', 'CS1K76\_To\_SP3', and 'SP3\_To\_CS1K76'. The 'CS1K76\_To\_SP3' profile is selected, and its configuration is shown in a table with columns: Header, Criteria, Replace Action, and Overwrite Value. The table contains three rows: 'To' with criteria 'IP/Domain', action 'Overwrite', and value '192.168.64.229'; 'Request-Line' with criteria 'IP/Domain', action 'Overwrite', and value '192.168.64.229'; and 'From' with criteria 'IP/Domain', action 'Overwrite', and value '10.10.98.111'. Buttons for 'Add', 'Rename', 'Clone', 'Delete', and 'Edit' are visible.

Header	Criteria	Replace Action	Overwrite Value
To	IP/Domain	Overwrite	192.168.64.229
Request-Line	IP/Domain	Overwrite	192.168.64.229
From	IP/Domain	Overwrite	10.10.98.111

Figure 59 - Topology Hiding Windstream

## 6.3. Domain Policies

The Domain Policies feature allows one to configure, apply, and manage various rule sets (policies) to control unified communications based upon various criteria of communication sessions originating from or terminating in the enterprise. These criteria can be used to trigger different policies which will apply on call flows, change the behavior of the call, and make sure the call does not violate any of the policies. There are default policies available to use, or one can create a custom domain policy.

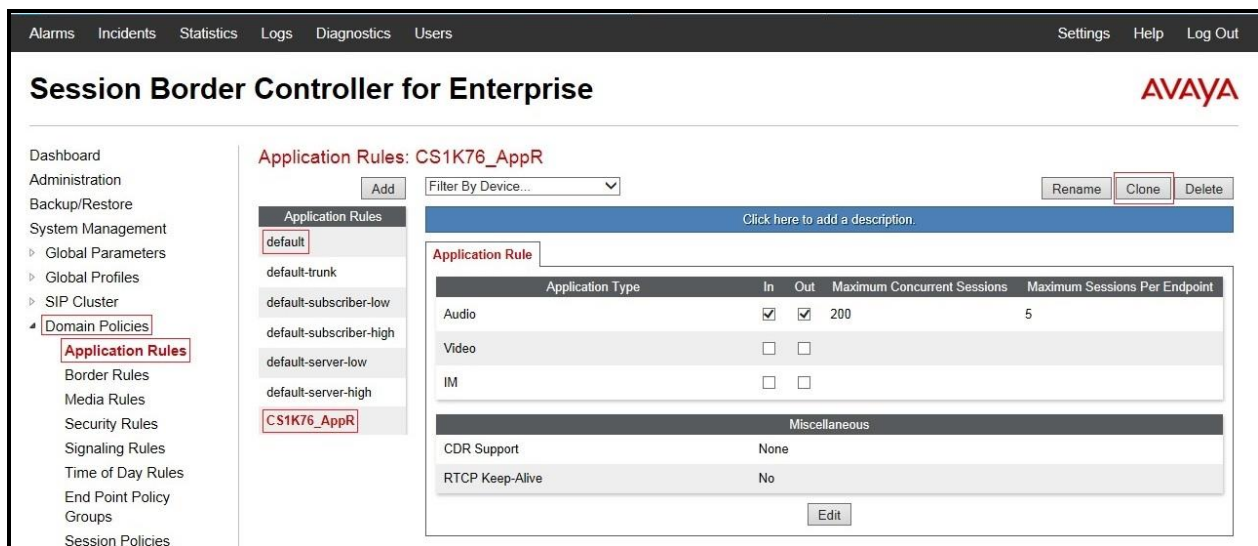


### 6.3.1. Create Application Rules

Application Rules allow one to define which types of SIP-based Unified Communications (UC) applications the Avaya SBCE security device will protect: voice, video, and/or Instant Messaging (IM). In addition, one can determine the maximum number of concurrent voice and video sessions so that the network will process to prevent resource exhaustion.

From the menu on the left-hand side, select **Domain Policies** → **Application Rules**.

- Select the **default** Rule
- Select **Clone** button
  - Name: **CS1K76\_AppR**
  - Click **Finish** (not shown).



**Figure 60 – Communication Server 1000 Application Rule**

From the menu on the left-hand side, select **Domain Policies** → **Application Rules**.

- Select the **default** Rule
- Select **Clone** button
  - Name: **SP3\_AppR**
  - Click **Finish** (not shown).

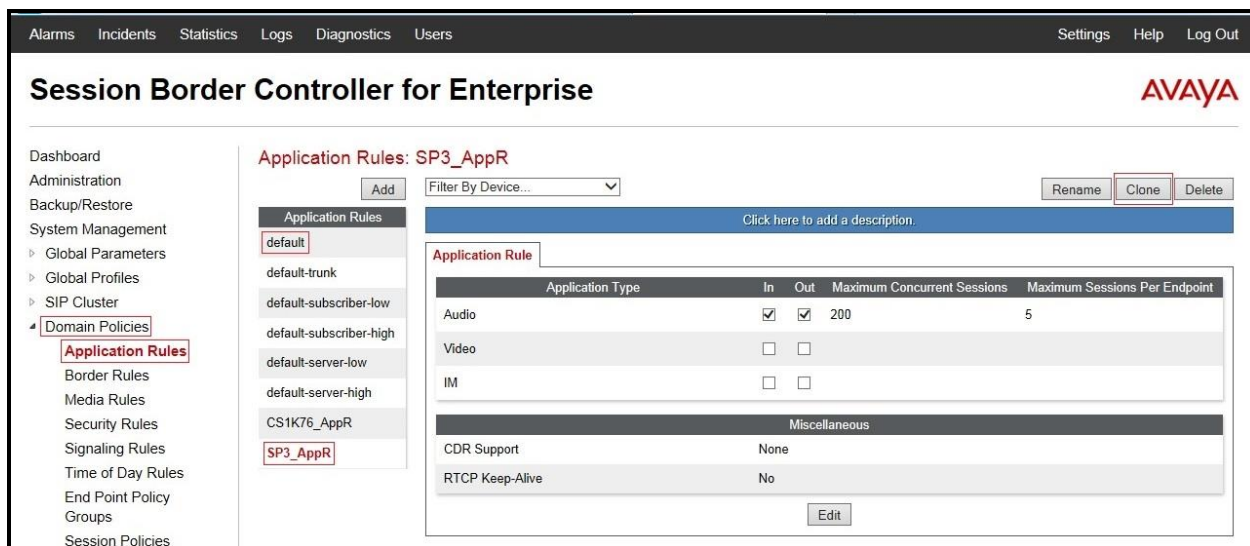


Figure 61 - Windstream Application Rule

### 6.3.2. Create Border Rules

Border Rules allow one to control NAT Traversal. The NAT Traversal feature allows one to determine whether or not call-flow through the DMZ needs to traverse a firewall and the manner in which pinholes will be kept open in the firewall to accommodate traffic.

From the menu on the left-hand side, select **Domain Policies** → **Border Rules**.

- Select the **default** Rule
- Select **Clone** button
  - Enter Clone Name: **CS1K76\_BorderR**
  - Click **Finish** (not shown).

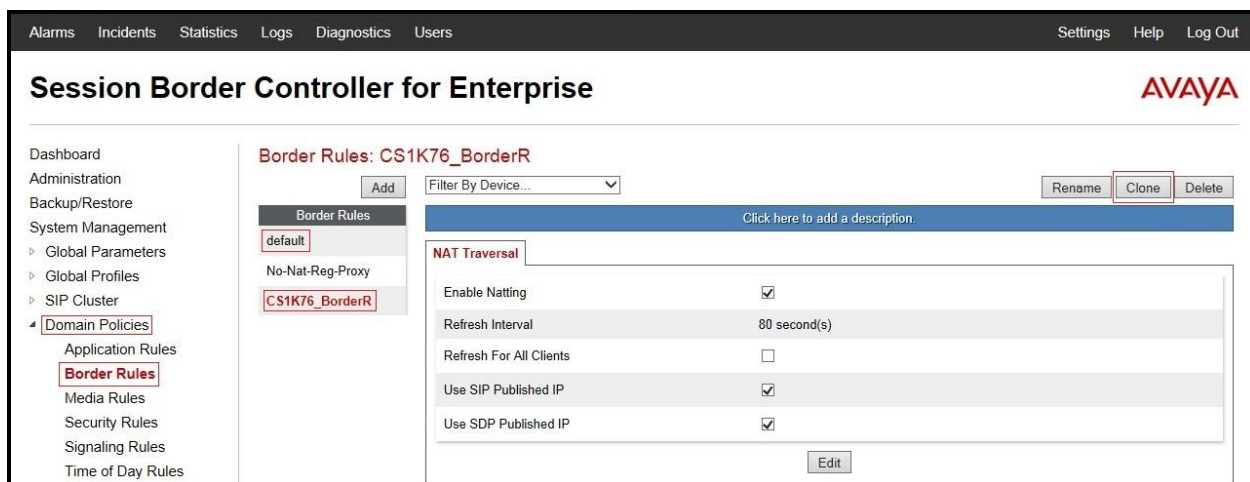
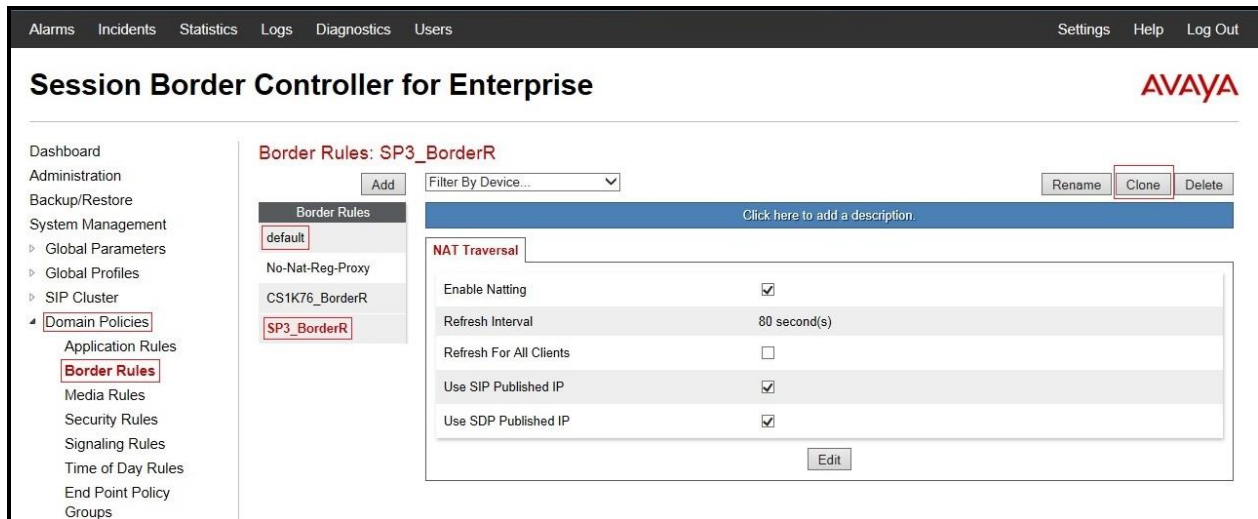


Figure 62 - Communication Server 1000 Border Rule

From the menu on the left-hand side, select **Domain Policies → Border Rules**.

- Select the **default** Rule
- Select **Clone** button
  - Enter Clone Name: **SP3\_BorderR**
  - Click **Finish** (not shown).



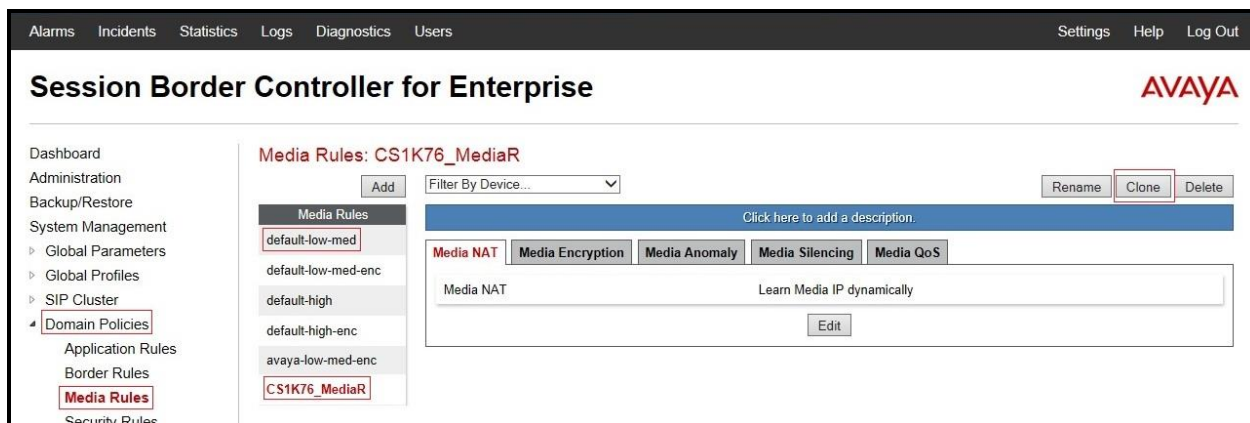
**Figure 63 - Windstream Border Rule**

### 6.3.3. Create Media Rules

Media Rules allow one to define RTP media packet parameters such as prioritizing encryption techniques and packet encryption techniques. Together these media-related parameters define a strict profile that is associated with other SIP-specific policies to determine how media packets matching these criteria will be handled by the Avaya SBCE security product.

From the menu on the left-hand side, select **Domain Policies → Media Rules**.

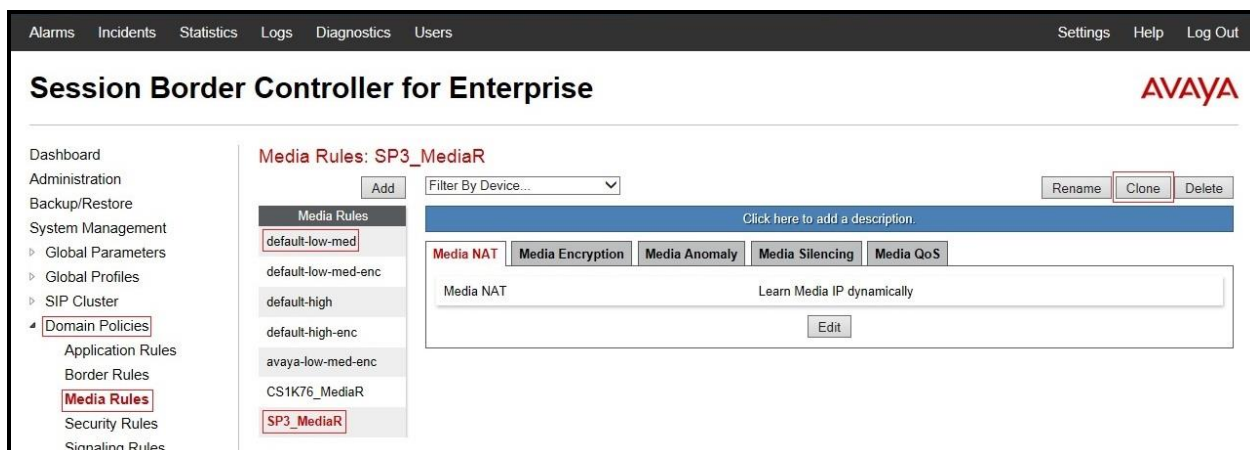
- Select the **default-low-med** Rule
- Select **Clone** button
  - Enter Clone Name: **CS1K76\_MediaR**
  - Click **Finish** (not shown).



**Figure 64 - Communication Server 1000 Media Rule**

From the menu on the left-hand side, select **Domain Policies** → **Media Rules**.

- Select the **default-low-med** Rule
- Select **Clone** button
  - Enter Clone Name: **SP3\_MediaR**
  - Click **Finish** (not shown).



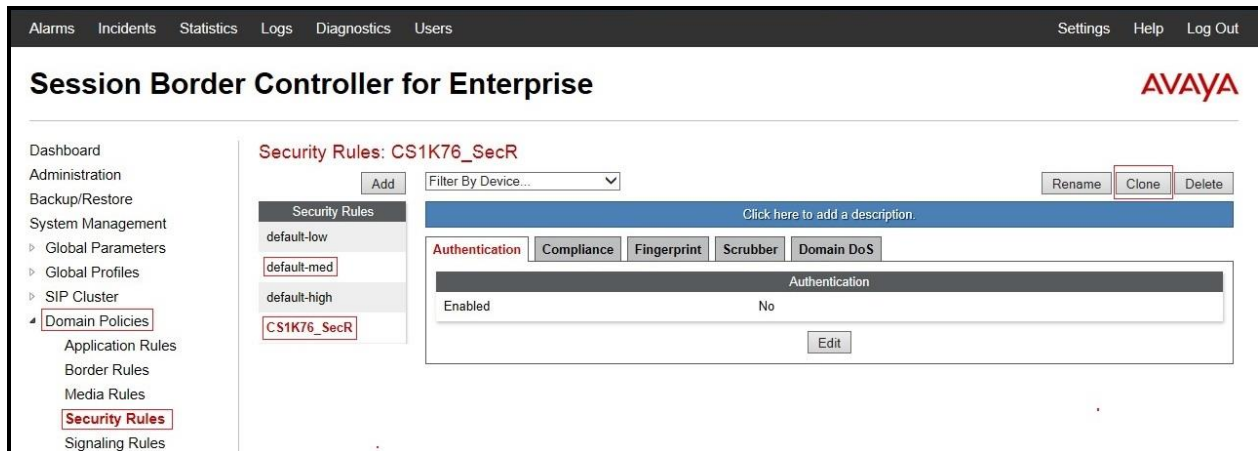
**Figure 65 – Windstream Media Rule**

### 6.3.4. Create Security Rules

Security Rules allow one to define which enterprise-wide VoIP and Instant Message (IM) security features will be applied to a particular call flow. Security Rules allows one to configure Authentication, Compliance, Fingerprinting, Scrubber, and Domain DoS. In addition to determining which combination of security features are applied, one can also define the security feature profile, so that the feature is applied in a specific manner to a specific situation. From the menu on the left-hand side, select **Domain Policies** → **Security Rules**.

- Select the **default-med** Rule
- Select **Clone** button

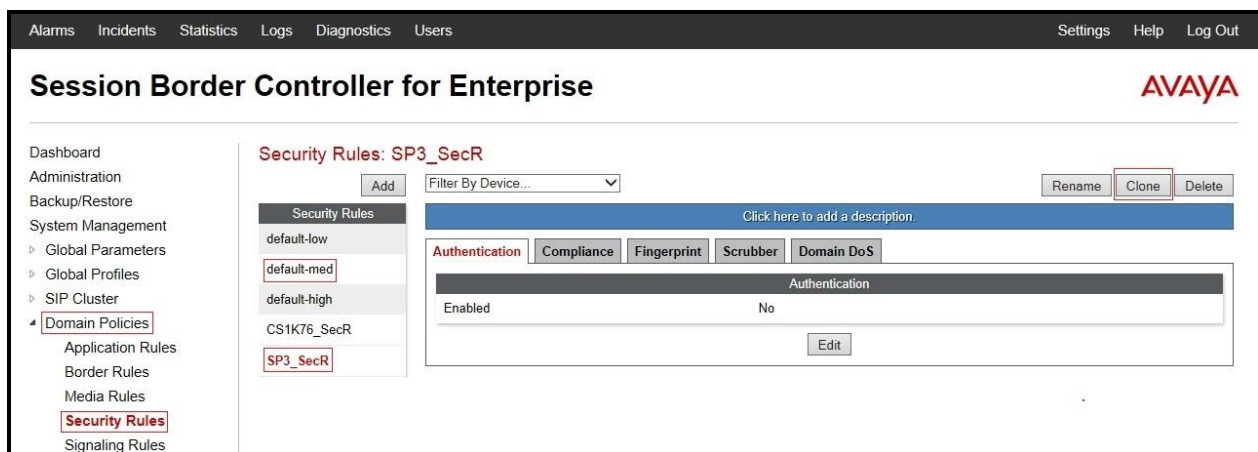
- Enter Clone Name: **CS1K76\_SecR**
- Click **Finish** (not shown).



**Figure 66 - Communication Server 1000 Security Rule**

From the menu on the left-hand side, select **Domain Policies** → **Security Rules**.

- Select the **default-med** Rule
- Select **Clone** button
  - Enter Clone Name: **SP3\_SecR**
  - Click **Finish** (not shown).



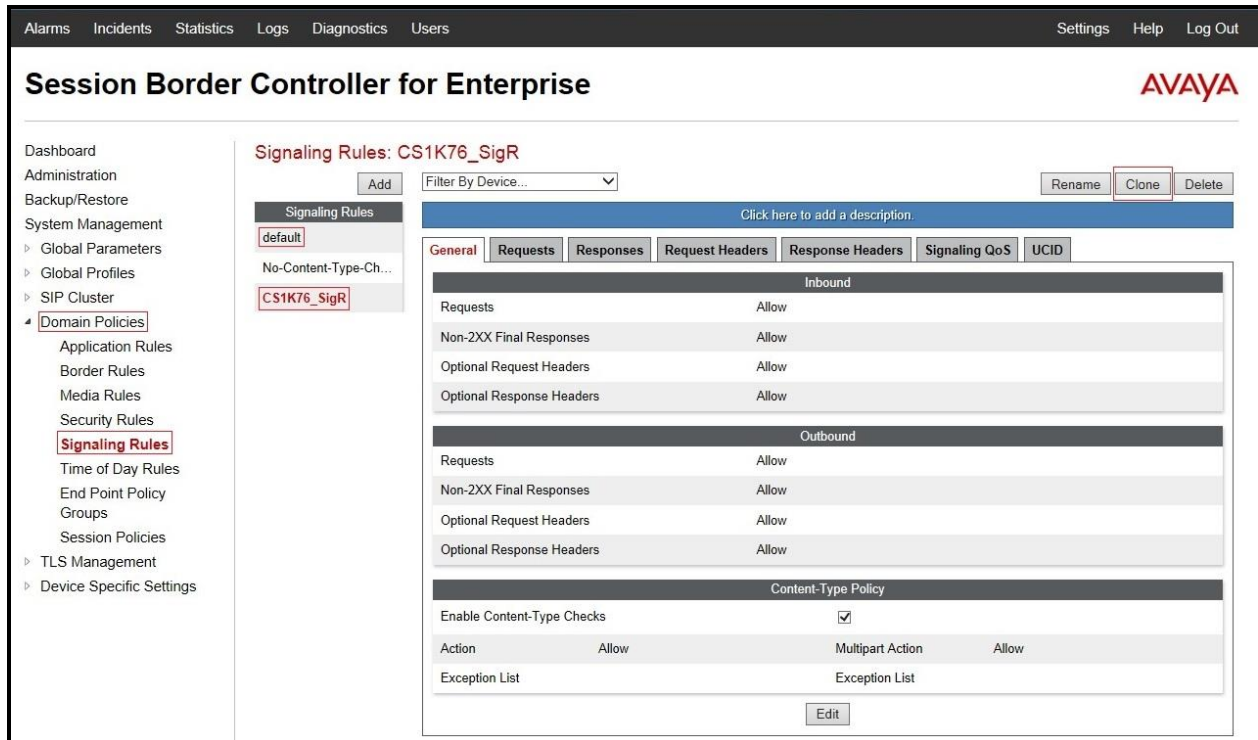
**Figure 67 - Windstream Security Rule**

### 6.3.5. Create Signaling Rules

Signaling Rules allow one to define the action to be taken (Allow, Block, Block with Response, etc.) for each type of SIP-specific signaling request and response message. When SIP signaling packets are received by the Avaya SBCE, they are parsed and “pattern matched” against the particular signaling criteria defined by these rules. Packets matching the criteria defined by the Signaling Rules are tagged for further policy matching.

From the menu on the left-hand side, select **Domain Policies** → **Signaling Rules**.

- Select the **default** Rule
- Select **Clone** button
  - Enter Clone Name: **CS1K76\_SigR**
  - Click **Finish** (not shown).



**Figure 68 - Communication Server 1000 Signaling Rule**

From the menu on the left-hand side, select **Domain Policies** → **Signaling Rules**.

- Select the **default** Rule
- Select **Clone** button
  - Enter Clone Name: **SP3\_SigR**
  - Click **Finish** (not shown).



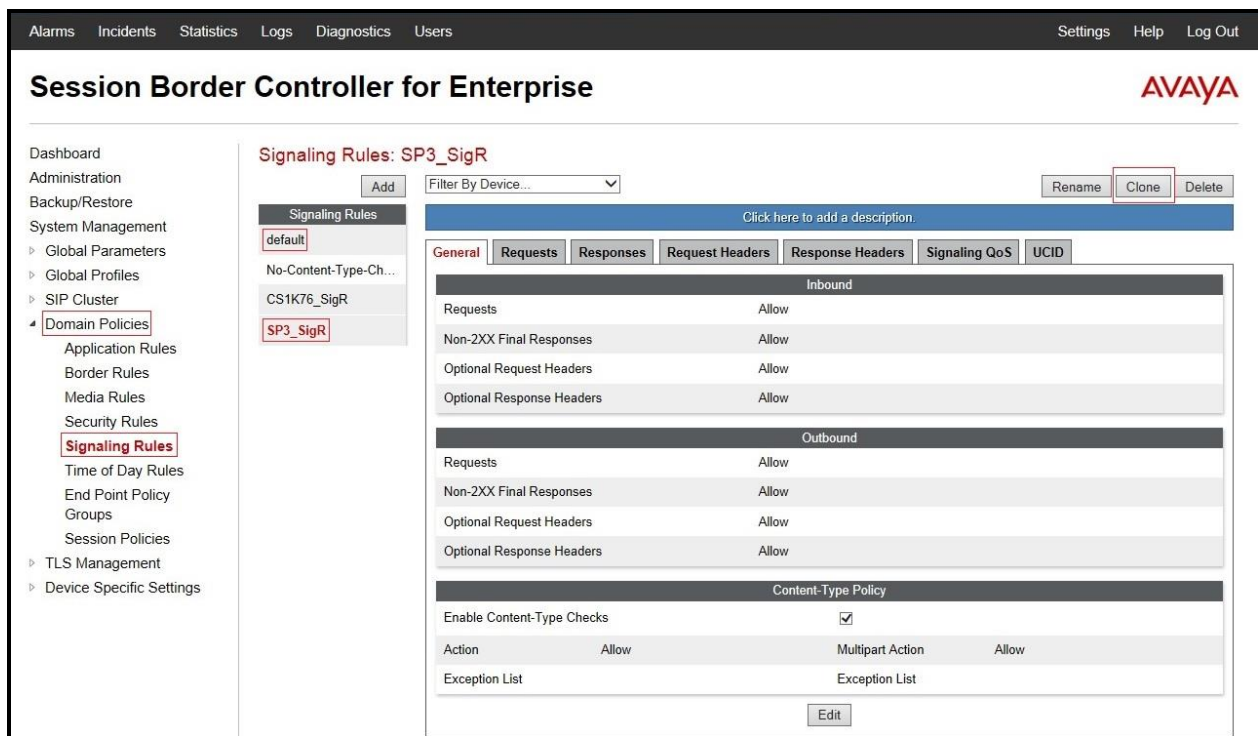


Figure 69 - Windstream Signaling Rule

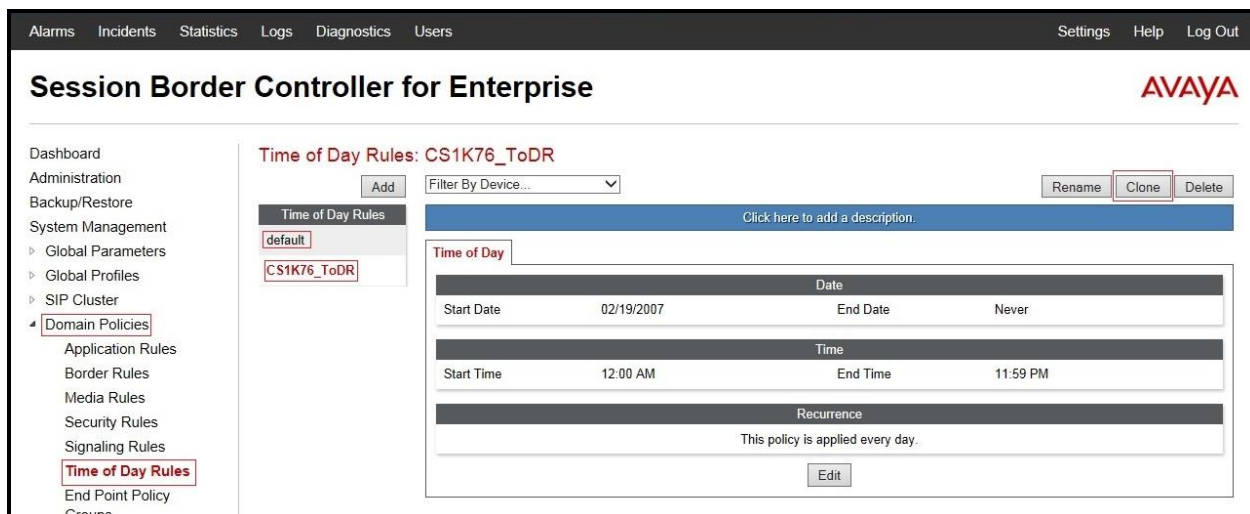
### 6.3.6. Create Time of Day Rules

A Time-of-day (ToD) Rule allows one to determine when the domain policy which is assigned to will be in effect. ToD Rules provide complete flexibility to fully accommodate the enterprise by, not only determining when a particular domain policy will be in effect, but also to whom it will apply, and for how long it will remain in effect.

From the menu on the left-hand side, select **Domain Policies** → **Time of Day Rules**.

- Select the **default** Rule
- Select **Clone** button
  - Enter Clone Name: **CS1K76\_ToDR**
  - Click **Finish** (not shown).

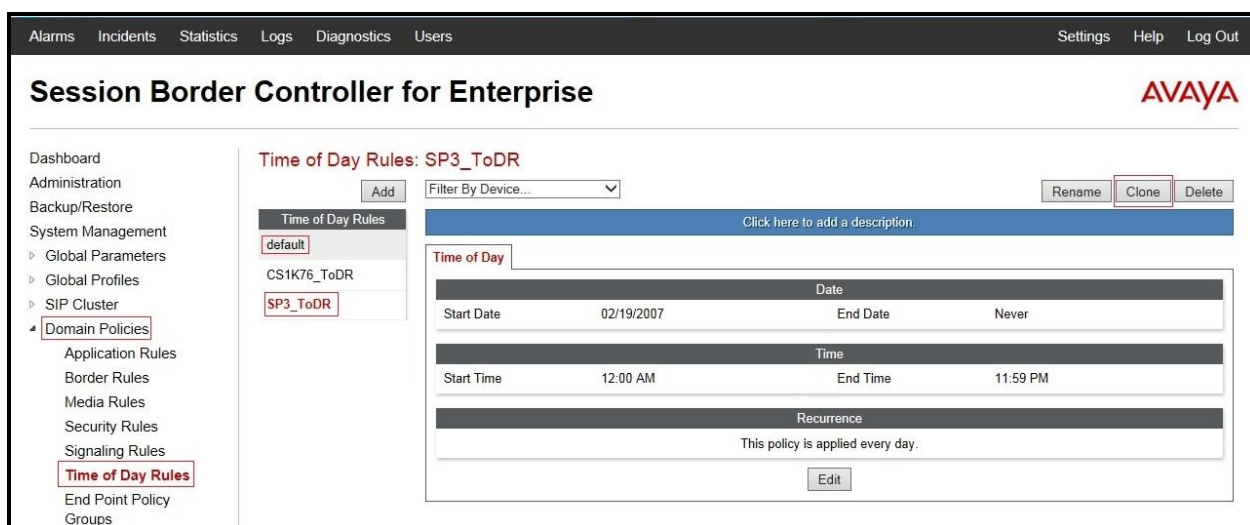




**Figure 70 - Communication Server 1000 Time of Day Rule**

From the menu on the left-hand side, select **Domain Policies** → **Time of Day Rules**.

- Select the **default** Rule
- Select **Clone** button
  - Enter Clone Name: **SP3\_ToDR**
  - Click **Finish** (not shown).



**Figure 71 - Windstream Time of Day Rule**

### 6.3.7. Create Endpoint Policy Groups

The End-Point Policy Group feature allows one to create Policy Sets and Policy Groups. A Policy Set is an association of individual, SIP signaling-specific security policies (rule sets): application, border, media, security, signaling, and ToD, each of which was created using the procedures contained in the previous sections.) A Policy Group is comprised of one or more Policy Sets. The purpose of Policy Sets and Policy Groups is to increasingly aggregate and simplify the application of Avaya SBCE security features to very specific types of SIP signaling messages traversing through the enterprise.

From the menu on the left-hand side, select **Domain Policies → End Point Policy Groups**.

- Select **Add**
- Enter **Group Name: CS1K76\_PolicyG**
  - **Application Rule: CS1K76\_AppR**
  - **Border Rule: CS1K76\_BorderR**
  - **Media Rule: CS1K76\_MediaR**
  - **Security Rule: CS1K76\_SecR**
  - **Signaling Rule: CS1K76\_SigR**
  - **Time of Day: CS1K76\_ToDR**
- Select **Finish** (not shown).

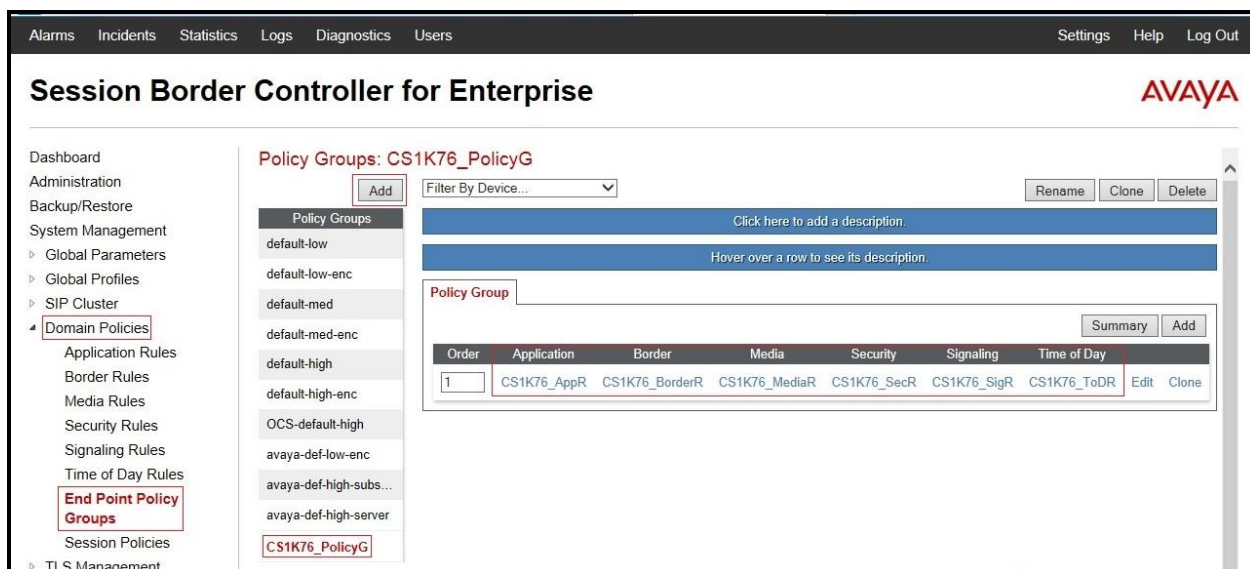


Figure 72 - Communication Server 1000 End Point Policy Group

From the menu on the left-hand side, select **Domain Policies → End Point Policy Groups**.

- Select **Add**
- Enter **Group Name: SP3\_PolicyG**
  - **Application Rule: SP3\_AppR**
  - **Border Rule: SP3\_BorderR**

- **Media Rule: SP3\_MediaR**
- **Security Rule: SP3\_SecR**
- **Signaling Rule: SP3\_SigR**
- **Time of Day: SP3\_ToDR**
- Select **Finish** (not shown).

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes links for Alarms, Incidents, Statistics, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header displays 'Session Border Controller for Enterprise' and the Avaya logo.

The left sidebar contains a navigation menu with the following items: Dashboard, Administration, Backup/Restore, System Management, Global Parameters, Global Profiles, SIP Cluster, Domain Policies (highlighted), Application Rules, Border Rules, Media Rules, Security Rules, Signaling Rules, Time of Day Rules, End Point Policy Groups (highlighted), Session Policies, TLS Management, and Device Specific Settings.

The main content area is titled 'Policy Groups: SP3\_PolicyG'. It features a list of policy groups on the left, including 'default-low', 'default-low-enc', 'default-med', 'default-med-enc', 'default-high', 'default-high-enc', 'OCS-default-high', 'avaya-def-low-enc', 'avaya-def-high-sub...', 'avaya-def-high-server', 'CS1K76\_PolicyG', and 'SP3\_PolicyG' (highlighted). The 'SP3\_PolicyG' group is selected, and its details are shown on the right.

The details for 'SP3\_PolicyG' include a table of associated rules:

Order	Application	Border	Media	Security	Signaling	Time of Day	
1	SP3_AppR	SP3_BorderR	SP3_MediaR	SP3_SecR	SP3_SigR	SP3_ToDR	Edit Clone

**Figure 73 - Windstream End Point Policy Group**

### 6.3.8. Create Session Policy

Session Policies allow users to define RTP media packet parameters such as codec types (both audio and video) and codec matching priority. Together these media-related parameters define a strict profile that is associated with other SIP-specific policies to determine how media packets matching these criteria will be handled by the Avaya SBCE security product.

- From the menu on the left-hand side, select **Domain Policies** → **Session Policies**.
- Select the **default** policy
- Select **Clone** button
  - Enter Clone Name: **SP3**
  - Click **Finish** (not shown).
- Click **Edit** button on **Media** tab
  - Check **Media Anchoring**
  - Select **Finish** (not shown).

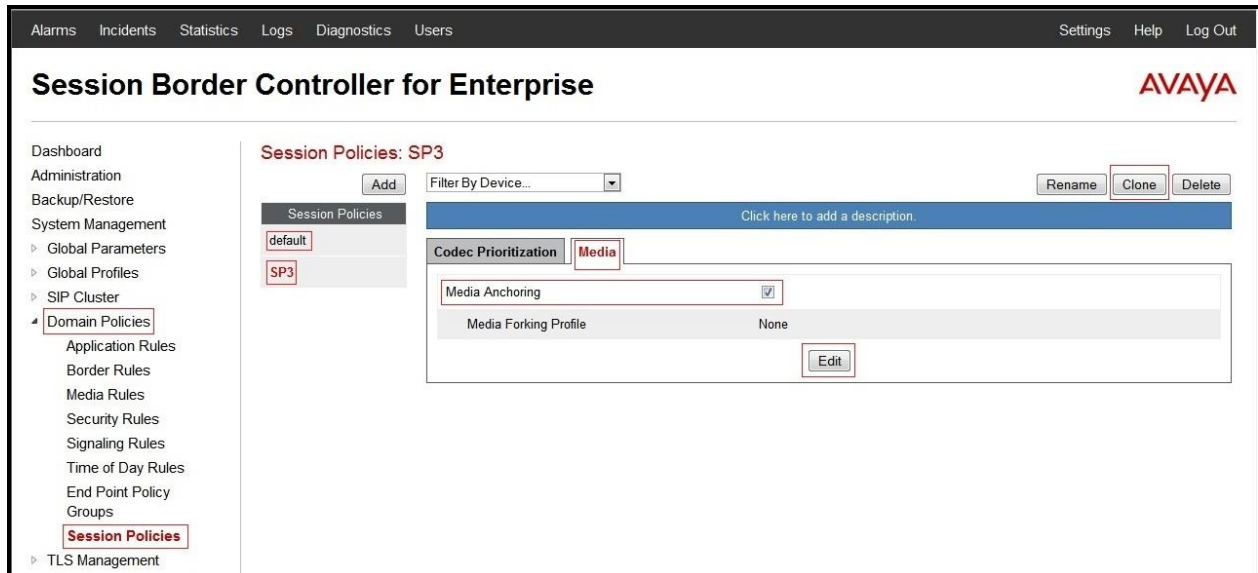


Figure 74 - Windstream Session Policy – Anchoring Media

## 6.4. Device Specific Settings

The Device Specific Settings feature for SIP allows one to view aggregate system information, and manage various device-specific parameters which determine how a particular device will function when deployed in the network. Specifically, one has the ability to define and administer various device-specific protection features such as Message Sequence Analysis (MSA) functionality, end-point and session call flows and Network Management.

### 6.4.1. Manage Network Settings

From the menu on the left-hand side, select **Device Specific Settings → Network Management**.

- Enter the **IP Address** and **Gateway Address** for both the Inside and the Outside interfaces:
  - **IP Address** for Inside interface: **10.10.98.13**; **Gateway: 10.10.98.1**
  - **IP Address** for Outside interface: **10.10.98.111**; **Gateway: 10.10.98.97**
- Select the physical interface used in the Interface column:
  - **Inside Interface: A1**
  - **Outside Interface: B1**

Alarms Incidents Statistics Logs Diagnostics Users Settings Help Log Out

### Session Border Controller for Enterprise

AVAYA

Dashboard  
Administration  
Backup/Restore  
System Management  
‣ Global Parameters  
‣ Global Profiles  
‣ SIP Cluster  
‣ Domain Policies  
‣ TLS Management  
‣ Device Specific Settings  
‣ **Network Management**  
Media Interface  
Signaling Interface  
Signaling Forking  
End Point Flows

#### Network Management: SBCE62

Devices  
SBCE62

Network Configuration Interface Configuration

Modifications or deletions of an IP address or its associated data require an application restart before taking effect. Application restarts can be issued from System Management.

Changes will not take effect until the interface is updated.

A1 Netmask: 255.255.255.192 A2 Netmask: B1 Netmask: 255.255.255.224 B2 Netmask:

Add Save Clear

IP Address	Public IP	Gateway	Interface	
10.10.98.13		10.10.98.1	A1	Delete
10.10.98.111		10.10.98.97	B1	Delete

Figure 75 - Network Management

- Select the **Interface Configuration** Tab.
- Toggle the State of the physical interfaces being used to **Enabled**.

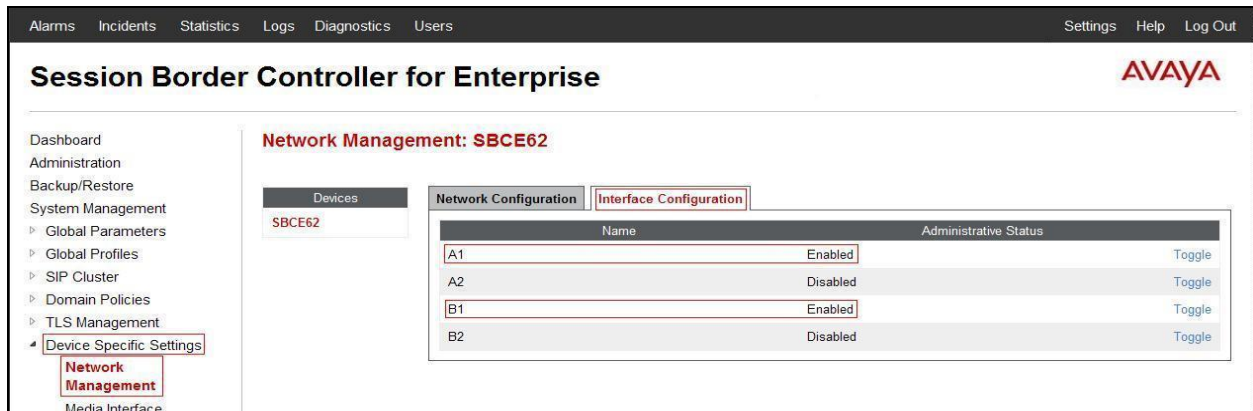


Figure 76 - Network Interface Status

### 6.4.2. Create Media Interfaces

Media Interfaces define the type of signaling on the ports. The default media port range on the Avaya can be used for both inside and outside ports.

From the menu on the left-hand side, **Device Specific Settings** → **Media Interface**.

- Select **Add**
  - **Name: InsideMedia**
  - **Media IP: 10.10.98.13** (Internal IP Address toward Communication Server 1000)
  - **Port Range: 35000 - 40000**
  - Click **Finish** (not shown)
- Select **Add**
  - **Name: OutsideMedia**
  - **Media IP: 10.10.98.111** (External IP Address toward Windstream trunk)
  - **Port Range: 35000 - 40000**
  - Click **Finish** (not shown).



Figure 77 - Media Interface



### 6.4.3. Create Signaling Interfaces

Signaling Interfaces define the type of signaling on the ports.

From the menu on the left-hand side, select **Device Specific Settings** → **Signaling Interface**.

- Select **Add**
  - **Name: InsideUDP**
  - **Media IP: 10.10.98.13** (Internal IP Address toward Communication Server 1000)
  - **UDP Port: 5060**
  - Click **Finish** (not shown).

From the menu on the left-hand side, select **Device Specific Settings** → **Signaling Interface**.

- Select **Add**
  - **Name: OutsideUDP**
  - **Media IP: 10.10.98.111** (External IP Address toward Windstream trunk)
  - **UDP Port: 5060**
  - Click **Finish** (not shown).

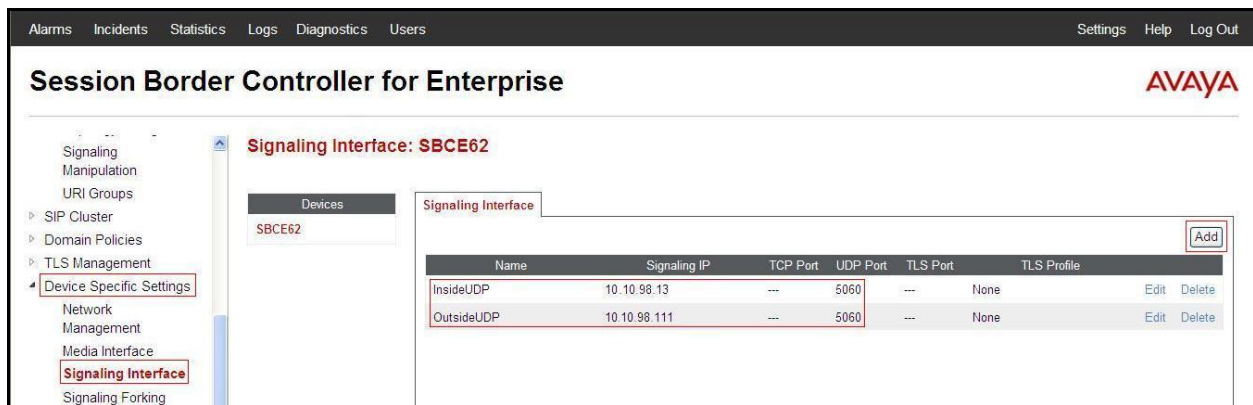


Figure 78 - Signaling Interface

### 6.4.4. Configuration Server Flows

Server Flows allow to categorize trunk-side signaling and to apply a policy.

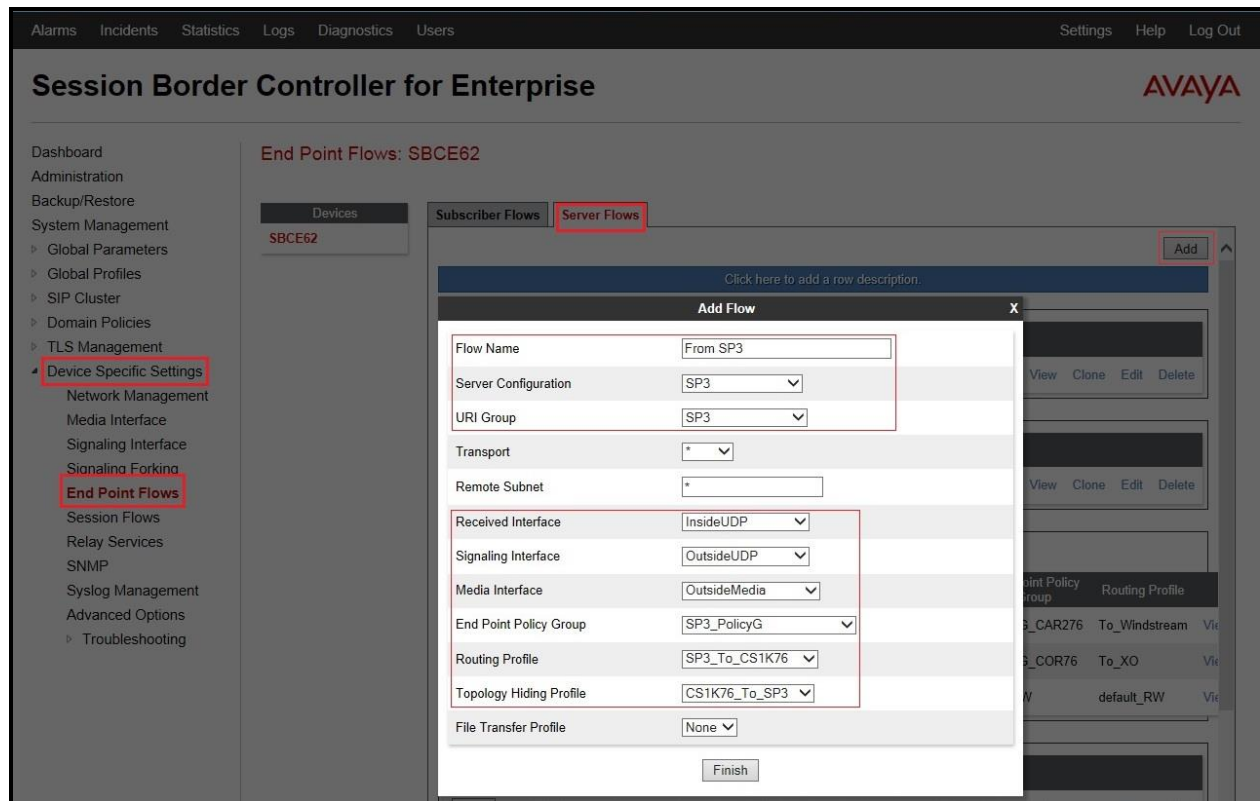
#### 6.4.4.1 Create End Point Flows – From Windstream

From the menu on the left-hand side, select **Device Specific Settings** → **End Point Flows**.

- Select the **Server Flows** Tab
- Select **Add**, enter **Flow Name: From SP3**
  - **Server Configuration: SP3**
  - **URI Group: SP3**
  - **Transport: \***
  - **Remote Subnet: \***
  - **Received Interface: InsideUDP**
  - **Signaling Interface: OutsideUDP**
  - **Media Interface: OutsideMedia**



- **End Point Policy Group: SP3\_PolicyG**
- **Routing Profile: SP3\_To\_CS1K76**
- **Topology Hiding Profile: CS1K76\_To\_SP3**
- Click **Finish**.

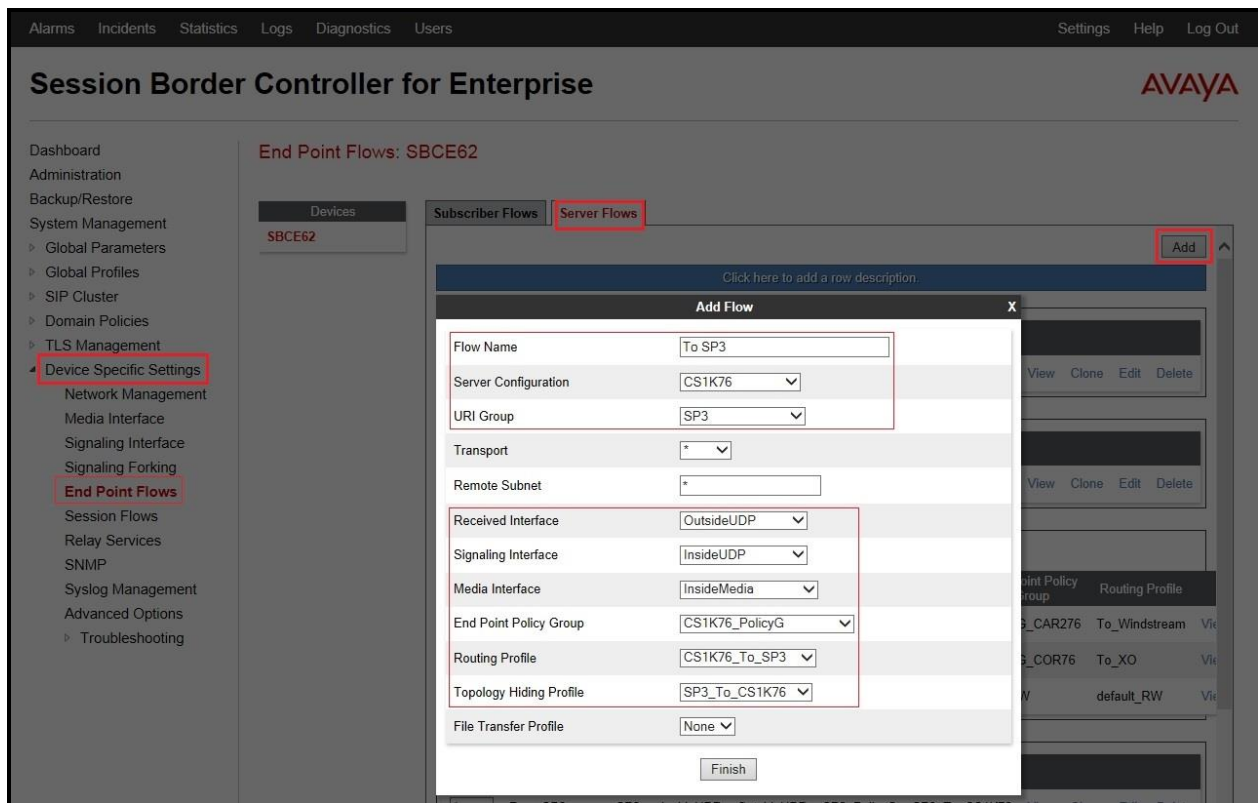


**Figure 79 - End Point Flows 1**

#### **6.4.4.2 Create End Point Flows – To Windstream**

From the menu on the left-hand side, select **Device Specific Settings → End Point Flows**.

- Select the **Server Flows** Tab
- Select **Add**, enter **Flow Name: To SP3**
  - **Server Configuration: CS1K76**
  - **URI Group: SP3**
  - **Transport: \***
  - **Remote Subnet: \***
  - **Received Interface: OutsideUDP**
  - **Signaling Interface: InsideUDP**
  - **Media Interface: InsideMedia**
  - **End Point Policy Group: CS1K76\_PolicyG**
  - **Routing Profile: CS1K76\_To\_SP3**
  - **Topology Hiding Profile: SP3\_To\_CS1K76**
  - Click **Finish**.



**Figure 80 - End Point Flows 2**

### 6.4.5. Create Session Flows

Session Flow determines the media (audio/video) sessions in order to apply the appropriate session policy.

- Select **Device Specific Settings** from the menu on the left-hand side
- Select the **Session Flows**
- Select **Add**
- Enter **Flow Name: SP3**
  - **URI Group#1: SP3**
  - **URI Group#2: SP3**
  - **Session Policy: SP3**
- Select **Finish** (not shown)

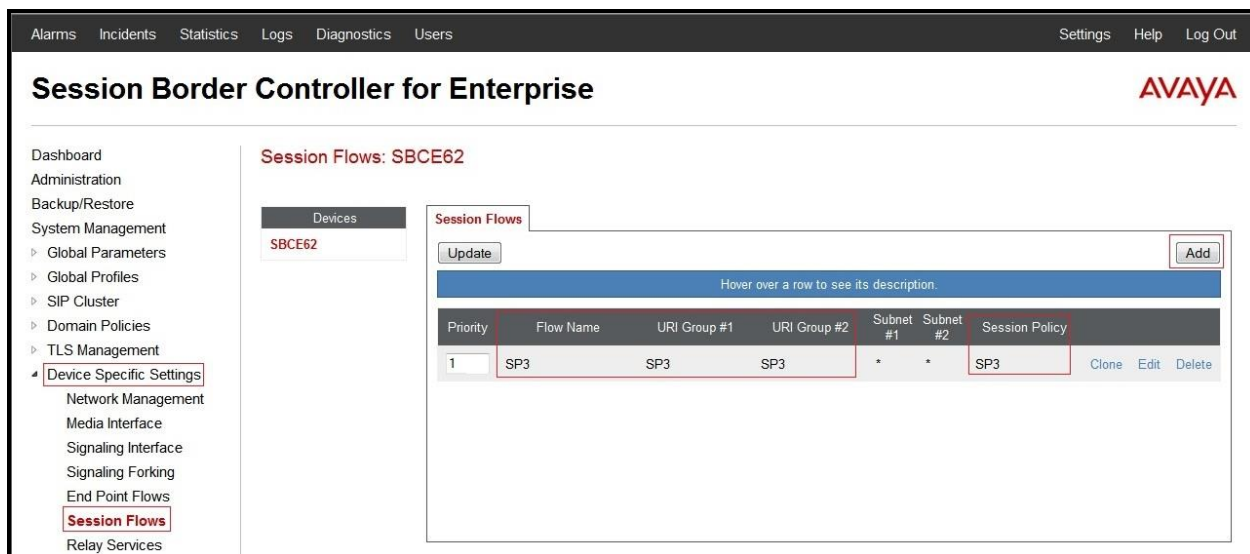


Figure 81 – Session Flows

## 7. Windstream SIP Trunking Service Configuration

Windstream is responsible for the network configuration of the Windstream SIP Trunking Service. Windstream will require that the customer provide the public IP address used to reach the Avaya SBCE public interface at the edge of the enterprise. Windstream will provide the IP address of Windstream's SIP proxy/SBC, IP addresses of media sources and Direct Inward Dialed (DID) numbers assigned to the enterprise. This information is used to complete configurations for Communication Server 1000, and the Avaya SBCE discussed in the previous sections.

The configuration between Windstream and the enterprise is a static configuration. There is no registration of the SIP trunk or enterprise users to the Windstream's network.

## 8. Verification Steps

The following steps may be used to verify the configuration.

### 8.1. General

Place an inbound call from a PSTN phone to an internal Avaya phone, answer the call, and verify that two-way speech path exists. Verify that the call remains stable for several minutes and disconnects properly.

### 8.2. Verification of an Active Call on Communication Server 1000

#### Active Call Trace (LD 80)

The following is an example of one of the commands available on the Communication Server 1000 to trace the DN for which the call is in progress or idle (1641). The call scenario involved PSTN phone number 6139675206 calling 6785931641 (which is translated to phone 1641).

- Login into Communication Server 1000 Signaling Server 10.10.97.177 with admin account and password.
- Issue a command “cslogin” to login on to the Communication Server 1000 Call Server.
- Log in to the Overlay command prompt, issue the command **LD 80** and then **trace 0 1641**.
- After the call is released, issue command **trac 0 1641** again to see if the DN is released back to idle state.

Below is the actual output of the Communication Server 1000 Call Server Command Line mode when the **1641** is in call state:

```
>ld 80
TRA000
.trac 0 1641

ACTIVE VTN 096 0 00 02

ORIG VTN 100 0 00 00 VTRK IPTI RMBR 100 1 INCOMING VOIP GW CALL
FAR-END SIP SIGNALLING IP: 10.10.98.13
FAR-END MEDIA ENDPOINT IP: 10.10.98.13 PORT: 37426
FAR-END SIP SIGNALLING IP: 10.10.98.13
FAR-END MEDIA ENDPOINT IP: 10.10.98.13 PORT: 37426
TERM VTN 096 0 00 02 KEY 0 SCR MARP CUST 0 DN 1641 TYPE 2002P2
SIGNALLING ENCRYPTION: INSEC
MEDIA ENDPOINT IP: 10.33.5.15 PORT: 5200
MEDIA PROFILE: CODEC G.711 MU-LAW PAYLOAD 20 ms VAD OFF
RFC2833: RXPT 101 TXPT 101 DIAL DN 1641
MAIN_PM ESTD
TALKSLOT ORIG 6 TERM 11
EES_DATA:
NONE
QUEU NONE
CALL ID 501 77
---- ISDN ISL CALL (ORIG) ----
CALL REF # = 484
BEARER CAP = VOICE
HLC =
CALL STATE = 10 ACTIVE
CALLING NO = 6139675206 NUM_PLAN:UNKNOWN TON:UNKNOWN ESN:UNKNOWN
CALLED NO = 6785931641 NUM_PLAN:UNKNOWN TON:UNKNOWN ESN:UNKNOWN
```

And this is the example after the call to 1641 is finished.

```
>ld 80
TRA000
.trac 0 1641
IDLE VTN 96 0 00 02 MARP
```

### SIP Trunk monitoring (LD 32)

Place a call inbound from PSTN (6139675206) to an internal device (6785931641). Then check the SIP trunk status by using LD 32, one trunk is BUSY.

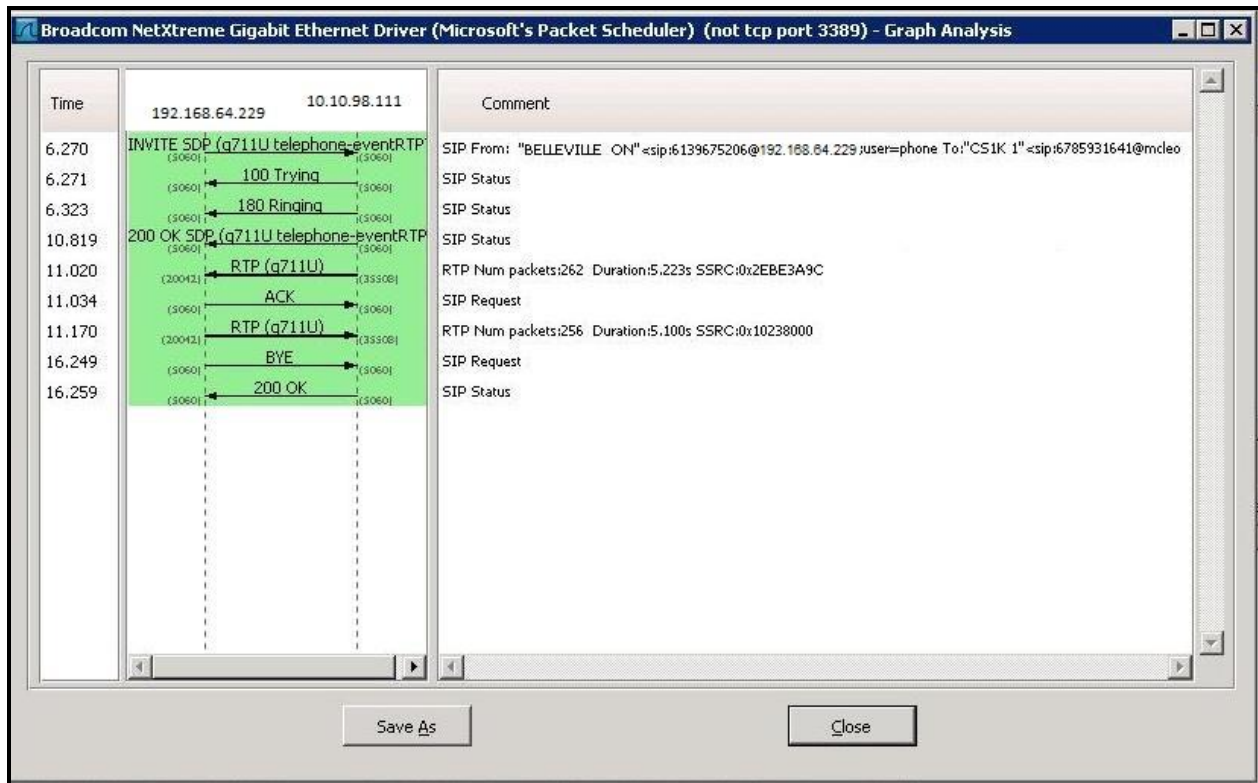
```
>ld 32
NPR000
.stat 100 0
091 UNIT(S) IDLE
001 UNIT(S) BUSY
000 UNIT(S) DSBL
000 UNIT(S) MBSY
```

After the call is released, check that SIP trunk status changed to the IDLE state.

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

### 8.3. Protocol Trace

Below is a wireshark trace of the same call scenario described in **Section 8.2**.



**Figure 82 – SIP Call Trace**

## 9. Conclusion

All of the test cases have been executed. Despite observations seen during the testing, as noted in **Section 2.2**, the test met the objectives outlined in **Section 2.1**. The Windstream SIP Trunk Service is considered **compliant** with Avaya Communication Server 1000 Release 7.6, and Avaya Session Border Controller for Enterprise Release 6.2.1 Q07.



## 10. References

This section references the documentation relevant to these Application Notes.

Product documentation for Avaya products, including the following, is available at:

<http://support.avaya.com/>

[1] Network Routing Service Fundamentals, Avaya Communication Server 1000, Release 7.6, Document Number NN43001-130, Issue 04.01, March 2013.

[2] IP Peer Networking Installation and Commissioning, Avaya Communication Server 1000, Release 7.6, Document Number NN43001-313, Issue 06.01, March 2013.

[3] Communication Server 1000E Overview, Avaya Communication Server 1000, Release 7.6, Document Number NN43041-110, Issue 06.01, March 2013.

[4] Unified Communications Management Common Services Fundamentals, Avaya Communication Server 1000, Release 7.6, Document Number NN43001-116, Issue 06.01, March 2013.

[5] Dialing Plans Reference, Avaya Communication Server 1000, Release 7.6, Document Number NN43001-283, Issue 06.01, March 2013.

[6] Product Compatibility Reference, Avaya Communication Server 1000, Release 7.6, Document Number NN43001-256, Issue 06.01 Standard, March 2013.

[7] Avaya Aura® System Manager Overview and Specification, Release 6.3, Issue 2, May 2013.

[8] Administering Avaya Session Border Controller for Enterprise, Release 6.2, Issue 2, May 2013.

[9] Avaya Session Border Controller for Enterprise Release notes, Release 6.2.1, Issue 5, December 2013.

Other resources:

[11] RFC 3261 SIP: Session Initiation Protocol, <http://www.ietf.org/>

[12] RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals, <http://www.ietf.org/>

## 11. Appendix A: SigMa Script

The following is the Signaling Manipulation script used in the configuration of the Avaya SBCE, **Section 6.2.6:**

```
within session "All"
{
act on message where %DIRECTION="INBOUND" and %ENTRY_POINT="AFTER_NETWORK"

    // Remove unwanted Diversion Header
    {
        remove(%HEADERS["Diversion"][1]);
    }
}

within session "All"
{
act on request where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
{
// Create the Diversion Header based on History Info

if (%HEADERS["History-Info"][1].regex_match("reason")) then
{

    %HEADERS["Diversion"][1] = "sip:dummy@dummy.com";

    %HEADERS["Diversion"][1].URI.SCHEME = %HEADERS["History-Info"][1].URI.SCHEME;
    %HEADERS["Diversion"][1].URI.USER = %HEADERS["History-Info"][1].URI.USER;
    %HEADERS["Diversion"][1].URI.HOST = %HEADERS["History-Info"][1].URI.HOST;
    %HEADERS["Diversion"][1].URI.PORT = %HEADERS["History-Info"][1].URI.PORT;

    %HEADERS["Diversion"][1].URI.PARAMS["reason"] = "unconditional";
    %HEADERS["Diversion"][1].URI.PARAMS["counter"] = "1";
    %HEADERS["Diversion"][1].URI.PARAMS["privacy"] = "off";
}

// Replace MIME by SDP

%HEADERS["Content-Type"][1].regex_replace("multipart/mixed;boundary=unique-boundary-1","application/sdp");

// Remove unwanted Headers

remove(%HEADERS["History-Info"][2]);
remove(%HEADERS["History-Info"][1]);
remove(%HEADERS["Alert-Info"][1]);
remove(%HEADERS["x-nt-e164-clid"][1]);
remove(%HEADERS["Remote-Address"][1]);

}
}
```

---

**©2014 Avaya Inc. All Rights Reserved.**

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at [devconnect@avaya.com](mailto:devconnect@avaya.com).