Avaya Solution & Interoperability Test Lab

Application Notes for Windstream SIP Trunking Service (Metaswitch) with Avaya Communication Server 1000 Release 7.6, Avaya Aura® Session Manager Release 6.3 and Avaya Session Border Controller for Enterprise Release 6.2 – Issue 1.1

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between Windstream SIP Trunking Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Communication Server 1000 Release 7.6, Avaya Aura® Session Manager Release 6.3, Avaya Session Border Controller for Enterprise Release 6.2 and various Avaya endpoints. This documented solution does not extend to configurations without Avaya Aura® Session Manager or Avaya Session Border Controller for Enterprise.

Windstream SIP Trunking Service provides PSTN access via a SIP Trunk between the enterprise and Windstream networks as an alternative to traditional PSTN trunks such as analog or ISDN-PRI. This approach generally results in lower cost for the enterprise.

Windstream is a member of the Avaya DevConnect Service Provider Program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing is conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between Windstream SIP Trunking Service (Windstream) and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Communication Server 1000 (CS1000) Release 7.6, Avaya Aura® Session Manager (Session Manager) Release 6.3, Avaya Session Border Controller for Enterprise (Avaya SBCE) Release 6.2 and various Avaya endpoints.

Windstream SIP Trunking Service referenced within these Application Notes is designed for enterprise business customers. Customers using Windstream SIP Trunking Service with the Avaya SIP-enabled enterprise solution are able to place and receive PSTN calls via a broadband WAN connection using the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as analog or ISDN-PRI.

2. General Test Approach and Test Results

Windstream is a member of the Avaya DevConnect Service Provider Program. The general test approach is to connect a simulated enterprise to Windstream via the public Internet and exercise the features and functionalities listed in **Section 2.1**.

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

To verify Windstream SIP Trunking interoperability, the following features and functionalities were covered during the compliance testing:

- Incoming PSTN calls to various phone types including UNIStim, SIP, digital and analog telephones at the enterprise. All incoming calls from PSTN are routed to the enterprise across the SIP Trunk from the service provider.
- Outgoing PSTN calls from various phone types including UNIStim, SIP, digital and analog telephones at the enterprise. All outgoing calls to PSTN are routed from the enterprise across the SIP Trunk to the service provider.
- Incoming and outgoing PSTN calls to/from 2050PC softphones.
- Dialing plans including local, long distance, international, outgoing toll-free, operator assisted calls, local directory assistance (411) calls, etc.
- Calling Party Name presentation and Calling Party Name restriction.
- Proper codec negotiation with G.711MU codec and G.729 codec.
- Proper early media transmission using G.711MU codec.
- Proper media transmission using G.711MU codec.
- Incoming and outgoing fax calls using G.711MU codec.

- DTMF tone transmission as out-of-band RTP events as per RFC 2833.
- Voicemail navigation for incoming and outgoing calls.
- Call Pilot voicemail hosted on the CS1000.
- Telephony features such as Hold and Resume, Call Waiting, Call Park, Call Transfer, Call Forward and Conferencing.
- Music on Hold.
- Off-net call transfer using subsequent INVITE method.
- Response to OPTIONS heartbeat.
- Response to incomplete call attempts and trunk errors.
- SIP Digest Authentication.
- Session Timers implementation.

Items that are not supported by Windstream on the test environment or not tested as part of the compliance testing, are listed as following:

- Inbound toll-free and outgoing emergency calls (E911) are supported but were not tested as part of the compliance testing.
- T.38 is not supported.
- Off-net calls transfer using REFER method is not supported.
- Call forward from PSTN back to PSTN is not supported.

2.2. Test Results

Interoperability testing of Windstream SIP Trunking Service with the Avaya SIP-enabled enterprise solution is completed with successful results for all test cases with the exception of the observations/limitations described below.

1. **For off-net blind transfer call**, **the calling PSTN does not hear ring back when the called PSTN is ringing**. When the CS1000 transfers off-net an incoming PSTN call back to PSTN, the transfer is successfully completed but after the transfer the calling PSTN does not hear ringback tone. This issue can be fixed by removing SDP in the 180 Ringing message coming to CS1000 from the Windstream, this can be done in Avaya SBCE in **Section 7.2.4.2** Server Interworking Profile for CS1000.
2. **No ringback tone on CS1000 UNIStim phone when it is blindly transferred by another CS1000 UNIStim phone to PSTN.** This is also a known issue in the CS1000. This issue can be fixed by removing SDP in the 180 Ringing message coming to CS1000 from the Windstream, this can be done in Avaya SBCE in **Section 7.2.4.2** Server Interworking Profile for CS1000.
3. **For off-net call transfer**, **Calling Party Name and Calling Party Number are not updated to PSTN parties**. When the CS1000 transfers off-net an incoming call back to PSTN, it does not update the true connected Calling Party Name and Calling Party Number to PTSN parties. The results are both PSTN parties still display Calling Party Name and Calling Party Number of the CS1000 extension. This is a known issue of the CS1000 when it interoperates with Windstream where the proprietary signaling of the CS1000 is not supported. This issue has low user impact, it is listed here simply as an observation.

4. **CS1000 UNIStim phone places an external call on hold then retrieves the held call**, **it causes Calling Party Number to change**. After retrieving a held external call, Calling Party Number previously displayed on the CS1000 UNIStim phone is replaced by "Route ACOD" – "Trunk Channel ID". This is a known behavior of the CS1000 with no resolution available at this time. This issue has low user impact, it is listed here simply as an observation.
5. **CS1000 UNIStim phone calls to an internal SIP phone which Call Forward All Calls to PSTN**, **the UNIStim phone does not display Calling Party Name and Number of the PSTN party**. After the call was successfully forwarded to PSTN, the PSTN party properly displayed DID number associated with the UNIStim or DID pilot number. However, the UNIStim phone still displayed local extension of the SIP phone which is not expected. It should display Calling Party Name and Number of the PSTN which is the true connected party. This is a known behavior of the CS1000 with no resolution available at this time. This issue has low user impact, it is listed here simply as an observation.
6. **CS1000 UNIStim phone calls to PSTN then blind transfers to an internal SIP phone, the SIP phone does not display Calling Party Name and Number of the PSTN party**. After the call was successfully transferred, the SIP phone displayed Calling Party Name and Number of the UNIStim which is not expected. It should display Calling Party Name and Number of the PSTN which is the true connected party. This is a known behavior of the CS1000 with no resolution available at this time. This issue has low user impact, it is listed here simply as an observation.
7. **MIME Multipart** in INVITE message sent from Avaya CS1000 is not supported by Windstream and it causes inbound call to PSTN dropped immediately after the call answered. Use an adaptation in Session Manager to remove MIME part in INVITE message to fix this issue. This adaptation is applied in Avaya SBCE SIP entity as configured in **Section 6.4** and **6.5**.
8. **Call Forward from PSTN back to PSTN** is not supported by Windstream (using Metaswitch). There is a workaround for this issue by applying a script in Signaling Manipulation of Avaya SBCE in **Section 7.2.5**.

2.3. Support

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

For technical support on Windstream SIP Trunking Service, please contact Windstream at http://www.windstream.com/Support/

3. Reference Configuration

Figure 1 illustrates the sample Avaya SIP-enabled enterprise solution connected to Windstream SIP Trunking Service (Vendor Validation Circuit) through the Internet. For confidentiality and privacy purposes, the actual public IP addresses and PSTN routable phone numbers used in the certification testing have been replaced with fictitious parameters throughout the Application Notes.

The Avaya SBCE is located at the edge of the enterprise network. The Avaya SBCE has two connection points, a public side connecting to Windstream via the Internet and a private side connecting to the enterprise network. All SIP and RTP traffic entering or leaving the enterprise network flow through the Avaya SBCE which can protect the enterprise against any outside SIP-based attacks. In the compliance testing, Windstream provided the service provider public IP address **64.xx.xxx.174**. This public IP address will be used for the public SIP traffic between the Avaya SBCE and Windstream. The Avaya lab was configured with a SIP domain **avayalab.com** for the enterprise, the Topology-Hiding feature of the Avaya SBCE (see **Section 7.2.3.1**) was used to adapt the enterprise SIP domain to the service provider SIP domains known to Windstream.

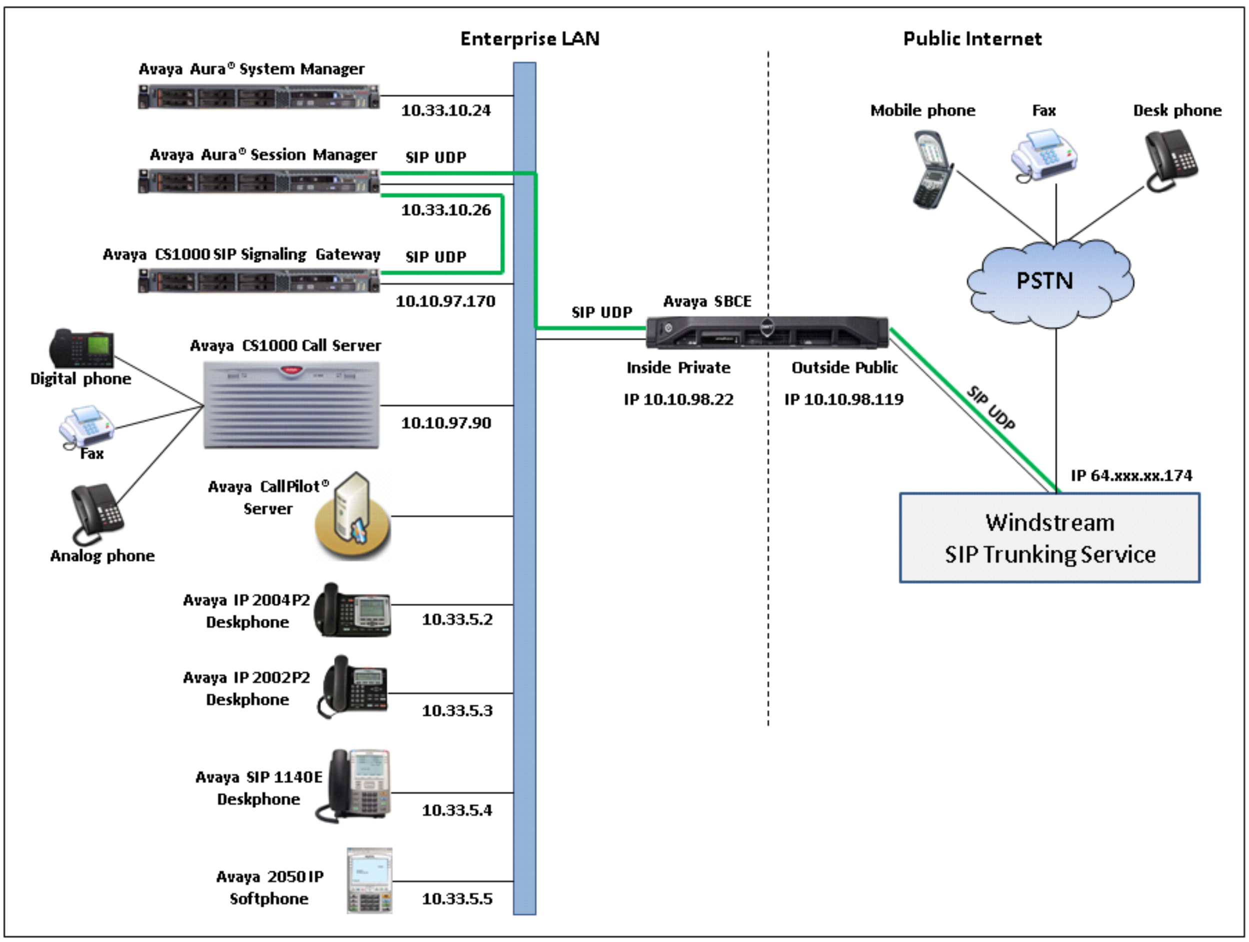


Figure 1: **Avaya IP Telephony Network connecting to Windstream SIP Trunking Service**

4. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Equipment/Software	**Release/Version**
Avaya CS1000 7.6 (CPPM)	• Call Server: 7.65 P GA plus latest DEPLIST Issue: 01 Release: 2013-12-17 (est) • SSG and SLG Server: 7.65.16 GA plus latest Service Pack 4 SP_7.6_4.ntl
Avaya Media Gateway Controller (MGC) Avaya DSP	• MGCCDC02 • DSP1AB07
Avaya Aura® Session Manager running on Avaya S8800 Server	6.3.4 – FP3 (6.3.4.4.1830)
Avaya Aura® System Manager running on Avaya S8800 Server	6.3.4 – FP3 (6.3.4.0.634014)
Avaya Call Pilot	05.00.41.141
Avaya IP Telephone	• 2002 p2: 0604DCO (UNIStim) • 2004 p2: 0604DCO (UNIStim) • 1140: 0625C8Q (UNIStim) • 1120: 0624C6Q (UNIStim) • 2007: 0621C8Q (UNIStim) • SIP 1140: SIP11x0e04.03.12.00
Avaya 2050PC softphone	4.3
Avaya Digital Telephone 3904	024
Avaya Analog Telephone	n/a
Avaya Session Border Controller for Enterprise (running on Dell R210 platform)	6.2.1 Q07
Windstream SIP Trunking Service Components	
Equipment/Software	**Release/Version**
Metaswitch	Version 6.0.3m5

Table 1: **Equipment and Software Tested**

5. Configure Avaya Communication Server 1000

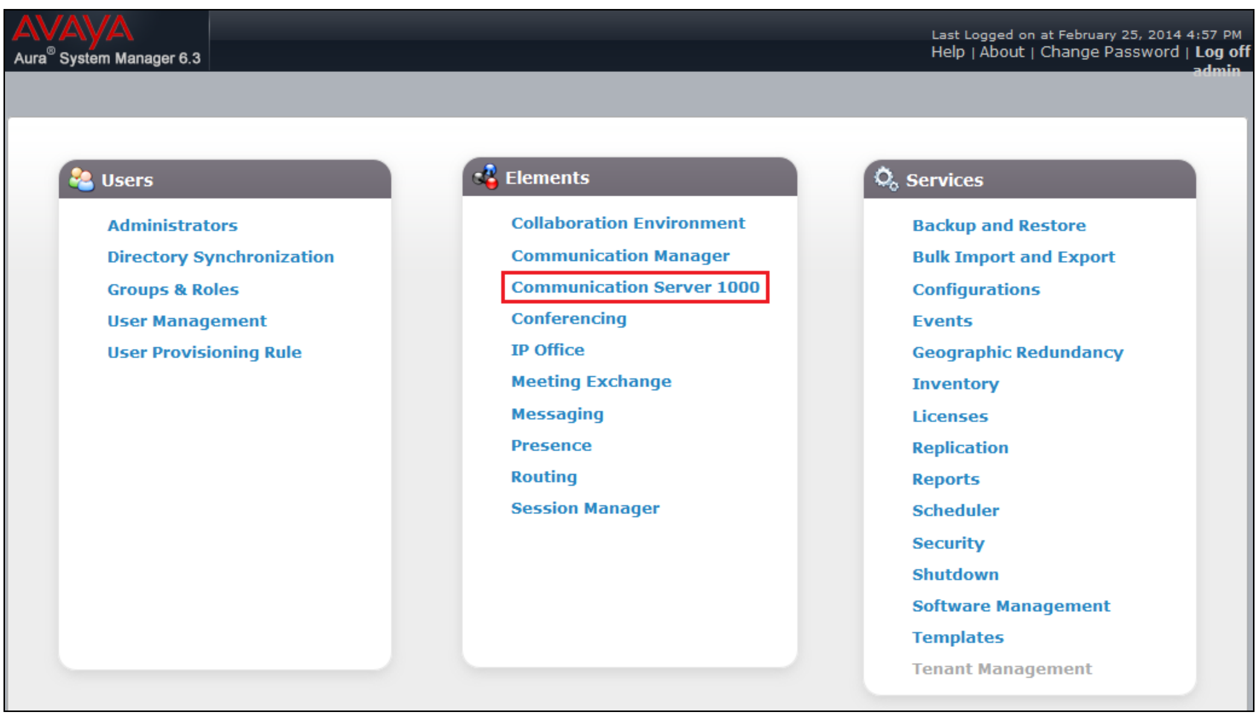
This section describes the procedure for configuring the CS1000 for inter-operating with Windstream. A two-way SIP Trunk was created between the CS1000 and Session Manager to carry traffic to and from the service provider respectively. Incoming calls flow from the Windstream networks to the Avaya SBCE to the CS1000 via Session Manager. Incoming calls into the CS1000 may undergo call treatments such as incoming digit translations and class of service restrictions. Outgoing calls to PSTN are first processed by the CS1000 for call treatments such as route selection and class of service. Once the CS1000 selects the proper SIP Trunk, the call is routed to the Avaya SBCE via Session Manager for egress to the Windstream networks.

These Application Notes assume the basic configuration has already been administered and it is not discussed here. For further information on the CS1000, see **References** in **Section 11**.

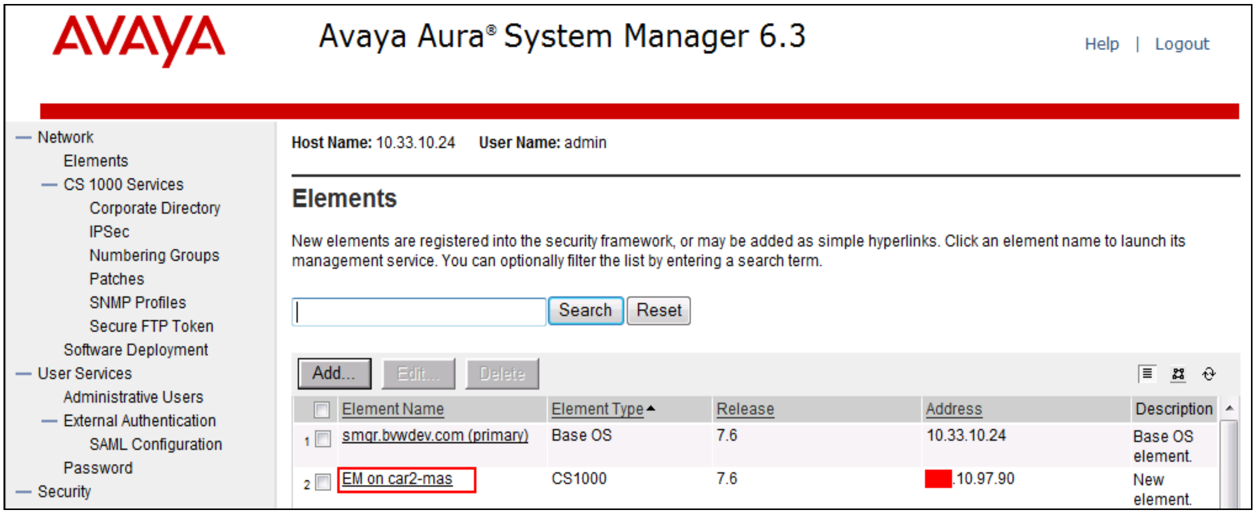
5.1. Log into the CS1000

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

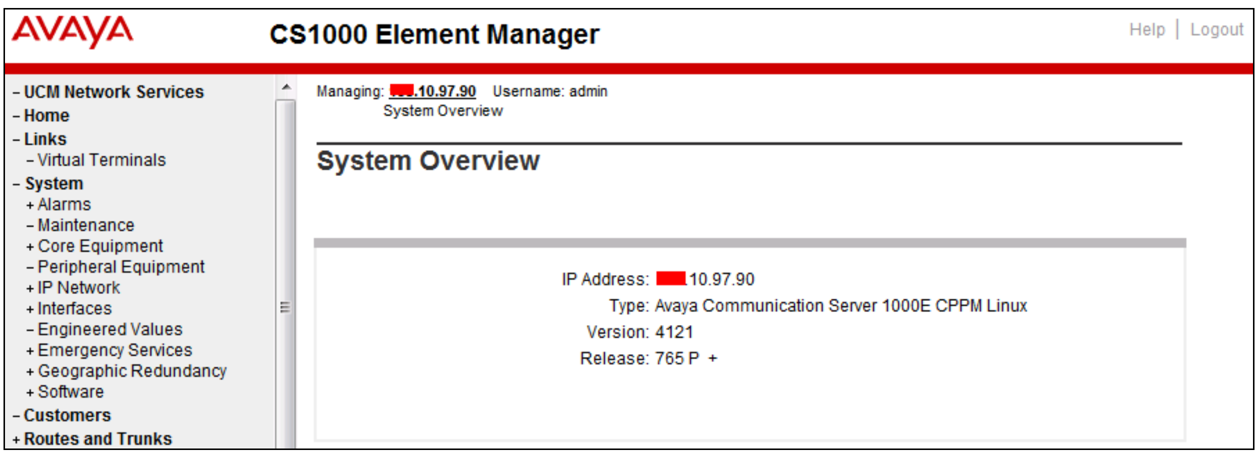
Since Release 7.6, CS1000 UCM is integrated with System Manager, depending on how the CS1000 system is deployed. For example, it might be deployed as standalone system and use its own UCM or within System Manager. In the compliance test configuration, UCM is accessed via System Manager. The screen below shows the System Manager home page with Communication Server 1000 entry in the **Elements** table. Click on the **Communication Server 1000** to access CS1000 UCM, the UCM webpage will be opened in the new window.



The **Avaya Unified Communications Management** is shown in the following screenshot. Click **Element Name** of the CS1000 Element as highlighted in the red box.



The following screenshot shows the CS1000 Element Manager **System Overview** page.



5.1.2. Log into Call Server Command Line Interface (CLI)

Using Putty, SSH to the IP address of the SIP Signaling Gateway (SSG) Server with the ***admin*** account then run the command ***cslogin*** and login with the appropriate admin account and password. The following screenshot are the logs.

```
login as: admin

               Avaya Inc. Linux Base  7.65
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@10.10.97.90's password:
Last login: Tue Oct  8 16:12:37 2013 from 10.10.98.86

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

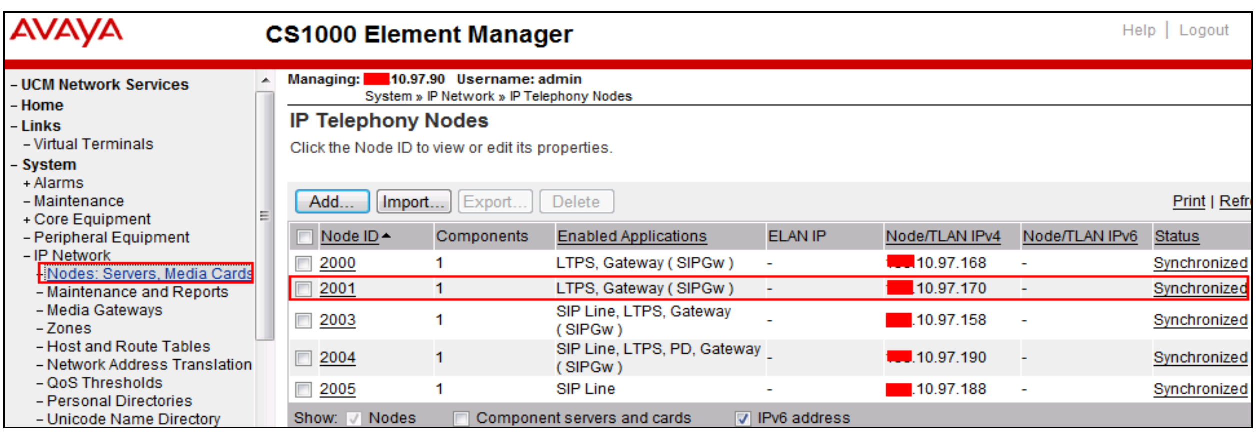
5.2. Administer Node IP Telephony

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

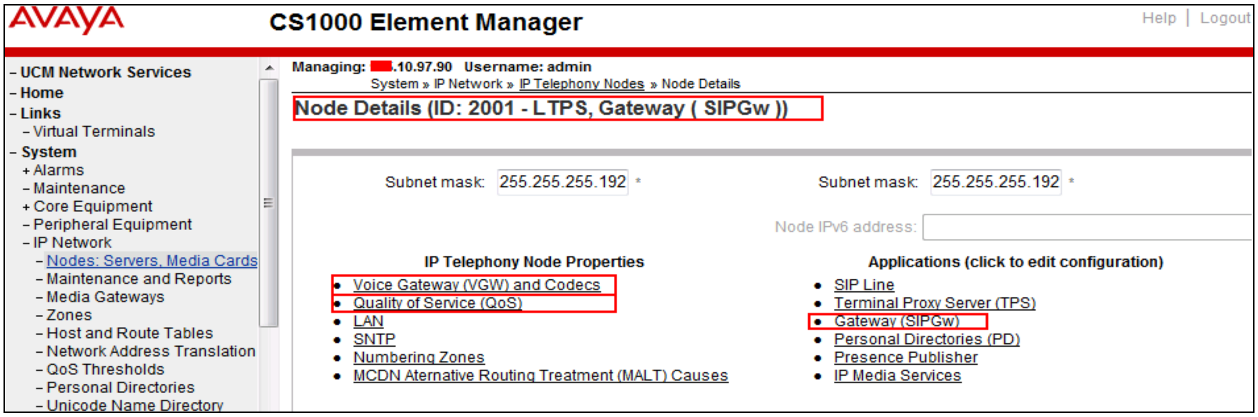
5.2.1. Obtain Node IP Address

These Application Notes assume the basic configuration has already been administered and that a Node has already been created. This section describes configuration steps for Node ID 2001.

To configure an IP Node, select **System → IP Network → Nodes: Servers, Media Cards**. In the **IP Telephony Nodes** page as shown in the screenshot below, click the **Node ID** of the CS1000.

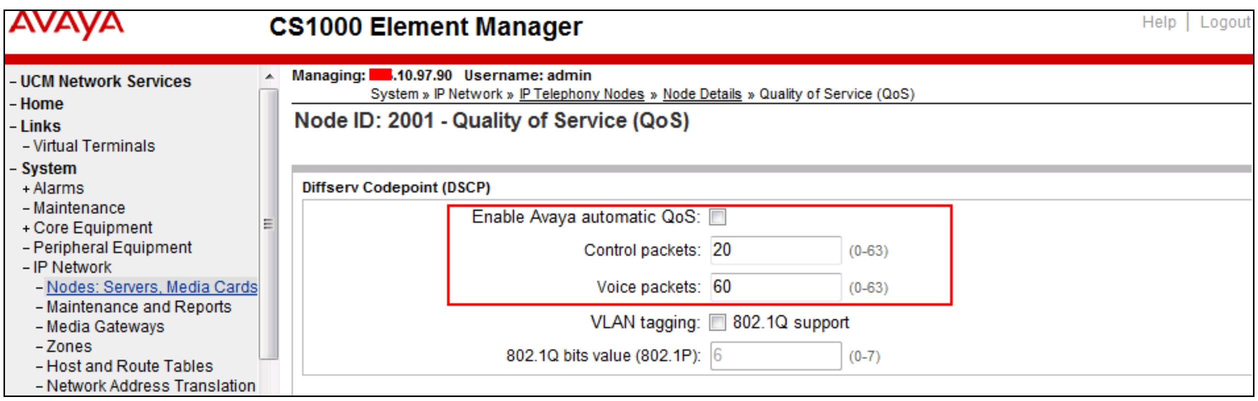


The **Node Details** page is shown in the screenshot below with the IP address of the Node ID 2001. The SIP Signaling Gateway uses the **Node IP Address** to connect to the Session Manager for the SIP Trunk to Windstream. The three highlighted in the screen shot below will be configured in next sections.



5.2.2. Administer Quality of Service (QoS)

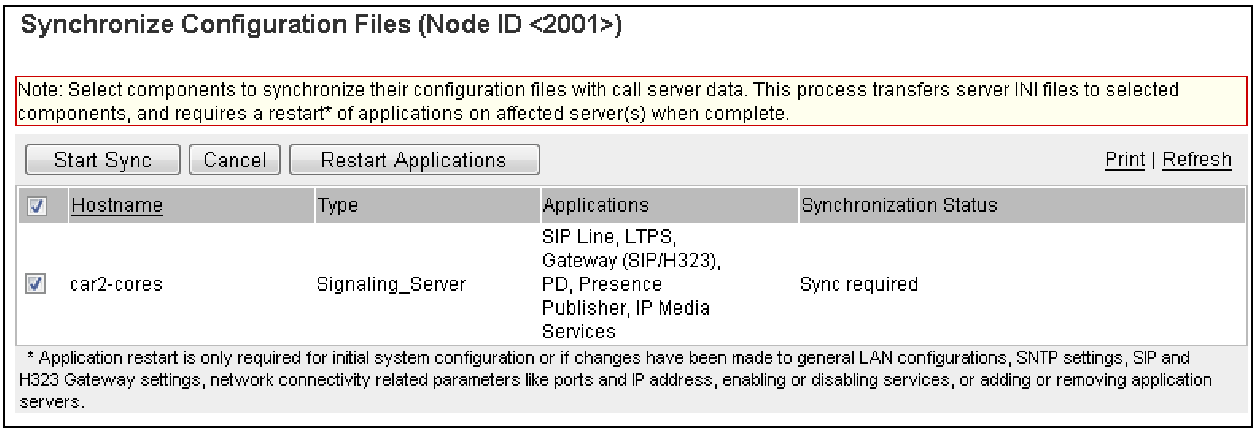
To configure the QoS, click **Quality of Service (QoS)** link in Node Details page shown in **Section 5.2.1**. Verify that the default Diffserv values were used as shown in the screenshot below, then click **Save** button (not shown).



5.2.3. Synchronize the new configuration

In order for the changes to take effect, the Node Details page needs to be saved and synchronized by following steps.

- Return to the **Node Details** page shown in **Section 5.2.1** and click **Save** button (not shown).
- The **Node Saved** screen is displayed. Click **Transfer Now** button (not shown).
- The **Synchronize Configuration Files** screen is displayed. Check the **Signaling Server** checkbox and click **Start Sync** button as shown below.
- When the synchronization completes, check the **Signaling Server** check box and click **Restart Applications** button (not shown).

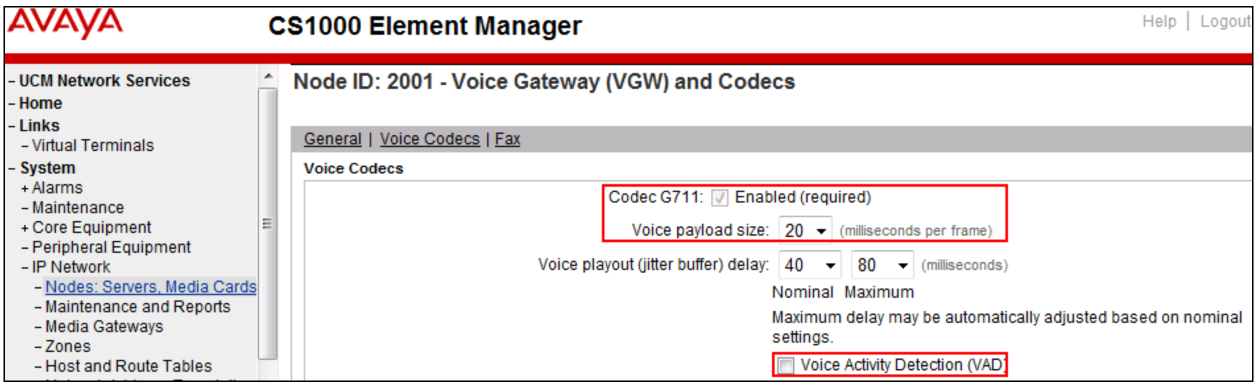


5.3. Administer Voice Codec

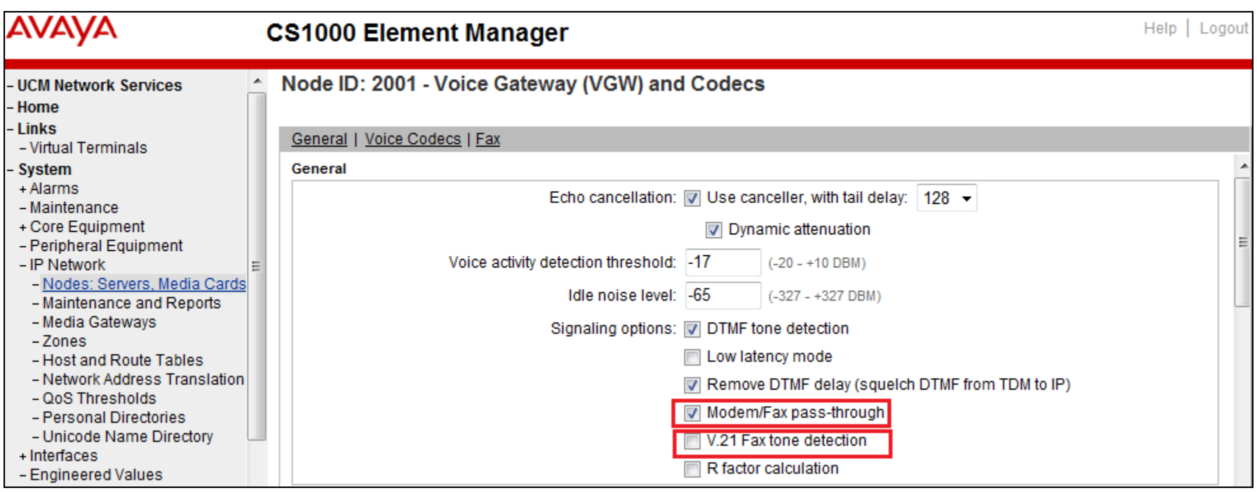
5.3.1. Enable Voice Codec, Node IP Telephony

To configure Voice Codec, select **IP Network → Nodes**: **Servers, Media Cards** from the left pane, and in the **IP Telephony Nodes** screen, select the **Node ID** of the CS1000 system. The **Node Details** screen is displayed as described in **Section 5.2.1**.

On the **Node Details** page (not shown), click on **Voice Gateway (VGW) and Codecs**. Windstream supports voice codec G.711 and G.729 (not shown), payload size 20 ms, with **Voice Activity Detection (VAD)** disabled. The following screenshot shows appropriate voice codec profile configured on the CS1000.



For Fax over IP, Windstream supports G.711 codec as default and does not support T.38. The following screenshot shows **Modem/Fax pass-through** is selected for **Node 2001**, this enables G.711 codec to be used for fax call between the CS1000 and Windstream. **Note**: The **V.21 Fax tone detection** should not be checked because **T.38 fax** is not supported.



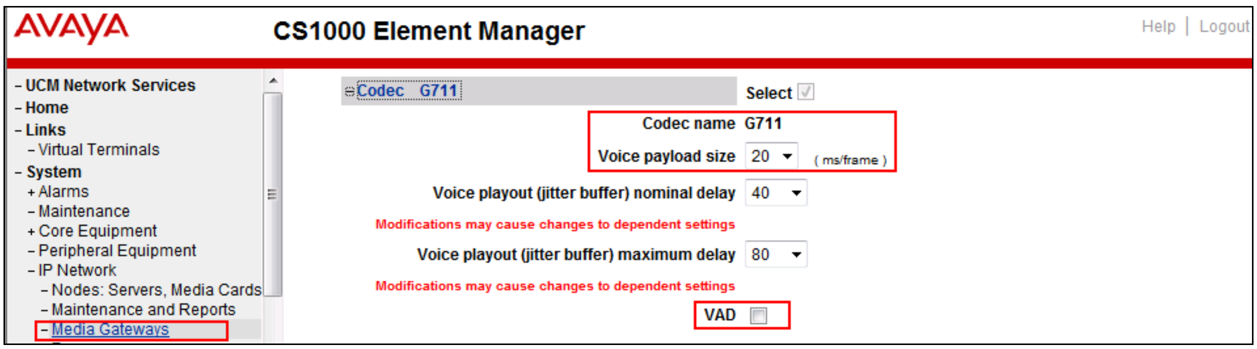
Click **Save** (not shown) then synchronize the new configuration (see **Section 5.2.3**).

5.3.2. Administer Voice Codec on Media Gateways

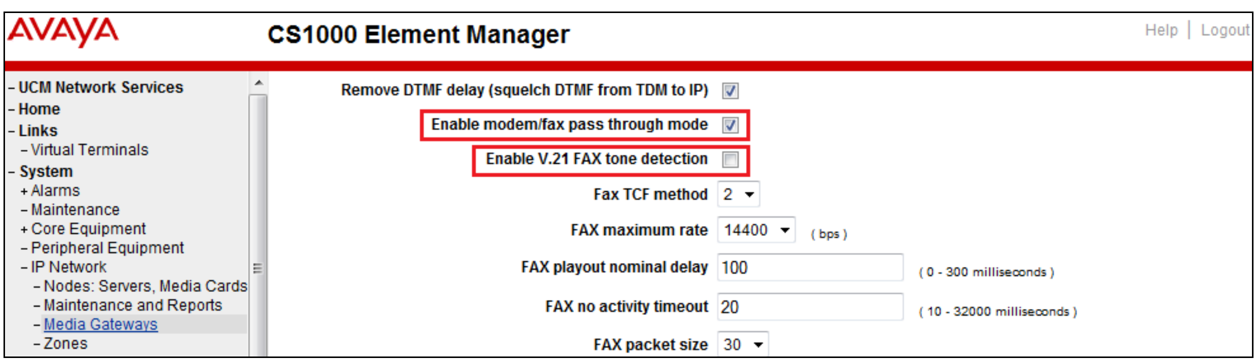
The CS1000 uses Media Gateways to support traditional analog and digital phones for voice calls over SIP Trunk. Media Gateways are also needed to support analog terminals to send fax over IP.

To configure Voice Codec on Media Gateways, from the left menu of the Element Manager page (not shown), select the **IP Network → Media Gateways** menu item. The Media Gateways page will appear (not shown). Click on the **MGC** which is located on the right of the page (not shown).

Windstream supports voice codec G.711, payload size 20 ms, with **VAD** disabled. The screenshot below shows appropriate codec profile configured for Media Gateways.



For Fax over IP, Windstream supports G.711 codec as default and does not support T.38. The following screenshot shows **Enable modem/fax pass through mode** is selected for Media Gateway, this enables G.711MU codec to be used for fax calls between the CS1000 and Windstream. **Note**: The **Enable V.21 FAX tone detection** should not be checked to disable T.38 fax capability on the Media Gateway.



5.4. Administer Zones and Bandwidth

This section describes the steps to create two zones: zone **10** for VGW and IP phone and zone **255** for SIP virtual trunk. The CS1000 uses zone configuration for bandwidth management purposes.

Windstream supports G.711 and G.729 codec on the test environment. In the sample configuration as shown in the screenshots below, the **MO** zone **10** and **VTRK** zone **255** were configured with **Strategy** of **Best Quality (BQ)** to allow the CS1000 to prioritize the G.711 codec for both voice and fax calls. **Note**: In the fax call scenario, the call has to be established with G.711 codec otherwise it will fail because the CS1000 cannot switch the codec over to G.711.

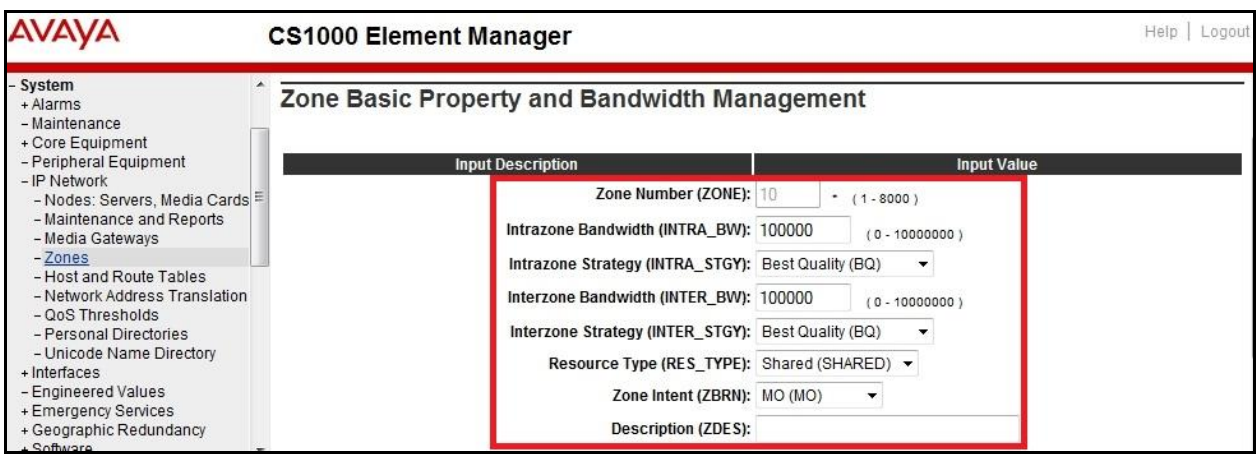
In general, a bandwidth zone is configured with parameters described as following:

- **INTRA_STGY**: Bandwidth configuration for local calls.
- **INTER_STGY**: Bandwidth configuration for the calls over the SIP Trunk.
  - **Note:** select **BQ** ifG.711 is first choice and G.729 is second choice.
  - **Note:** select **BB** if G.729 is first choice and G.711 is second choice.
- **MO**: The zone type which is used for IP phones and VGW.
- **VTRK**: The zone type which is used for the SIP Trunk.

5.4.1. Create Zone for VGW and IP phones

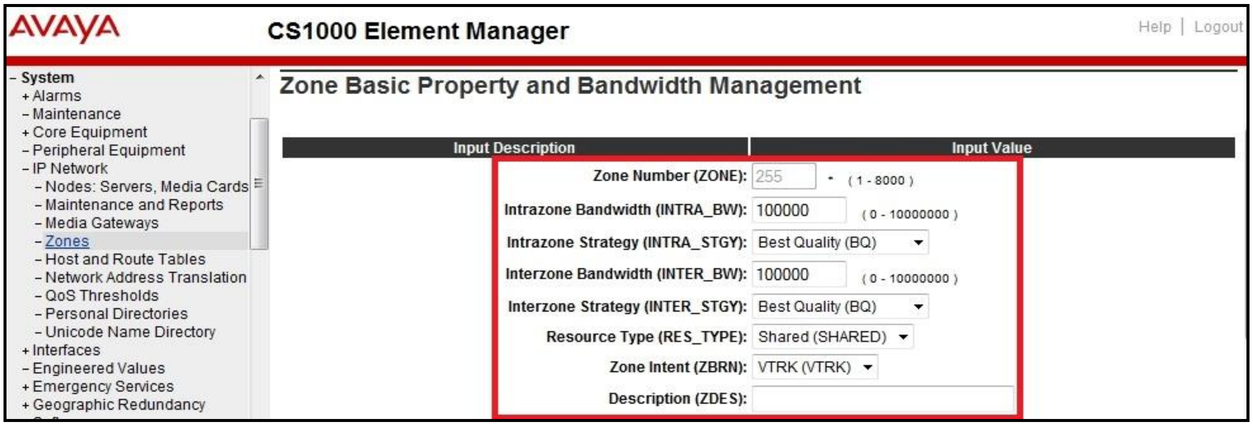
To create a MO zone **10** for VGW and IP phone, select **IP Network → Zones** from the left pane and then configure as following:

- Click **Bandwidth Zones** link (not shown).
- In **Bandwidth Zones** screen, click **Add** button (not shown).
- In the **Add Bandwidth Zone** screen, click on **Zone Basic Property and Bandwidth Management**, select the values as shown (in red box) in the screenshot below and click on the **Submit** button (not shown).



5.4.2. Create Zone for virtual SIP Trunk

Follow **Section 5.4.1** to create a VTRK zone **255** for the virtual trunk. The difference is in the **Zone Intent (ZBRN)** field, select **VTRK** for virtual trunk as shown in the screenshot below then click **Submit** button (not shown).



5.5. Administer SIP Trunk Gateway

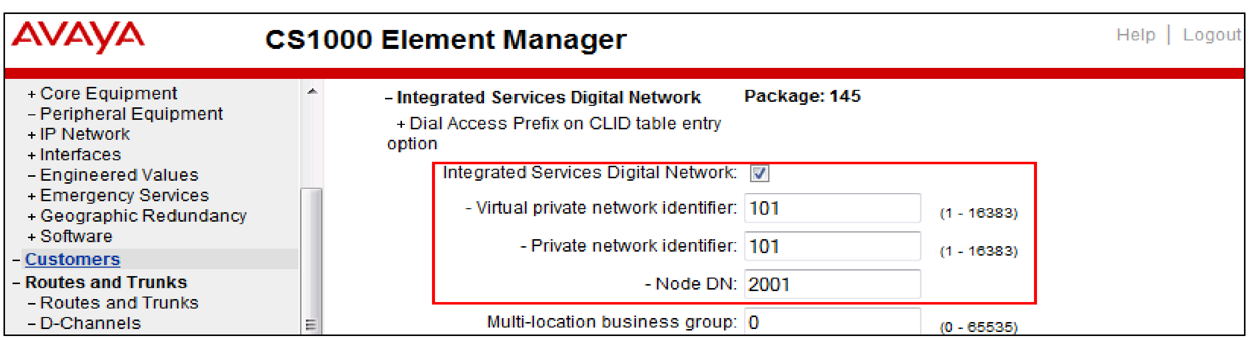
This section describes the steps for establishing a SIP Trunk between the CS1000 SIP Signaling Gateway (SSG) and Session Manager.

5.5.1. Integrated Services Digital Network (ISDN)

To support ISDN in the SIP trunk, select **Customers** in the left pane. The **Customers** screen is displayed (not shown). Click on the link associated with the appropriate customer, in this case is **01**. The system can support more than one customer with different network settings and options. The **Customer 01 Edit** page will appear (not shown). Select the **Feature Packages** option from this page (not shown).

The screen is populated with a list of **Feature Packages**. Select **Integrated Services Digital Network** to edit its parameters. The screen is populated with **Integrated Services Digital Network** parameters as follows.

- Virtual private network identifier: Enter a valid value, e.g. **101**.
- Private network identifier: Enter a valid value, e.g. **101**.
- Node DN: Enter the Node DN, e.g. **2001**.



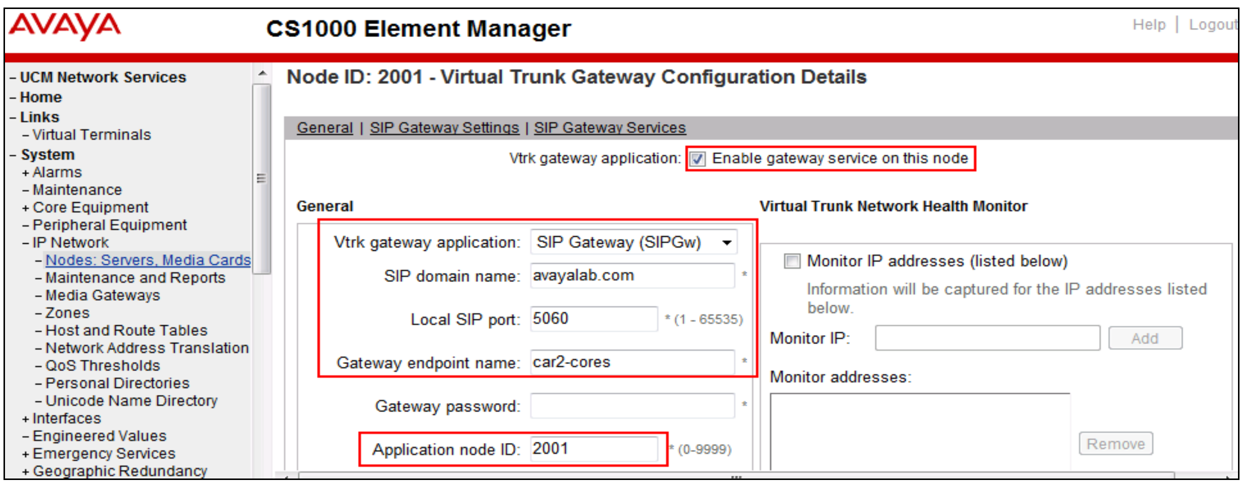
Retain the default values for all remaining fields. Scroll down to the bottom of the screen then click **Save** button (not shown).

5.5.2. Administer SIP Trunk Gateway to the Session Manager

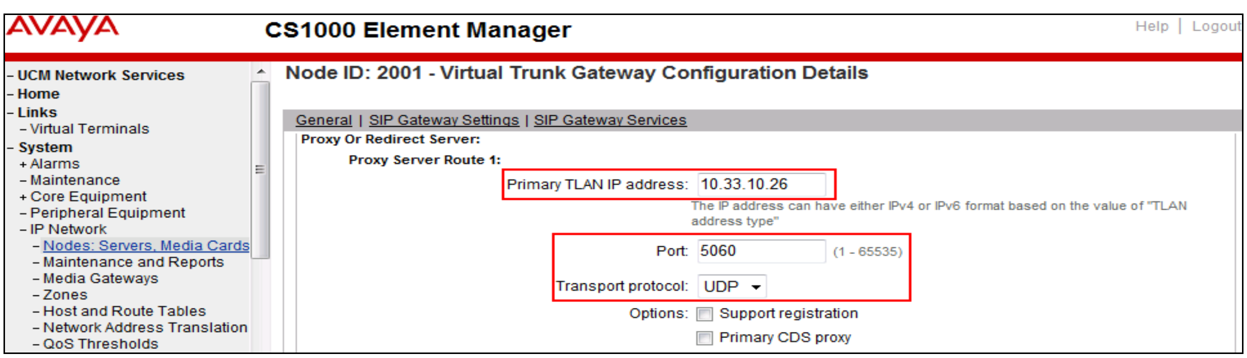
To configure SIP Trunk Gateway, select **IP Network → Nodes: Servers, Media Cards** configuration from the left pane, and in the **IP Telephony Nodes** screen, select the **Node ID 2001**. The **Node Details** screen is displayed as shown in **Section 5.2.1**.

On the **Node Details** screen, select **Gateway (SIPGw)** (not shown). Check on the check box "**Enable gateway service on this node**". Under **General** tab of the **Virtual Trunk Gateway Configuration Details** screen, enter the following values which are highlighted in red boxes as shown in screenshot below.

- **Vtrk gateway application**: Select **SIP Gateway (SIPGw)**.
- **SIP domain name**: An enterprise SIP Domain name, .e.g. **avayalab.com**.
- **Local SIP port**: A port open to receive SIP traffic, .e.g. **5060**.
- **Gateway endpoint name**: A descriptive name for SIP Gateway, .e.g. **car2-cores**.
- **Application node ID**: An available node ID, .e.g. **2001**.



Click on the **SIP Gateway Settings** tab, under **Proxy or Redirect Server**, enter the IP address 10.33.10.26 of Session Manager as shown in the screenshot below, and retain the default values for the remaining fields.



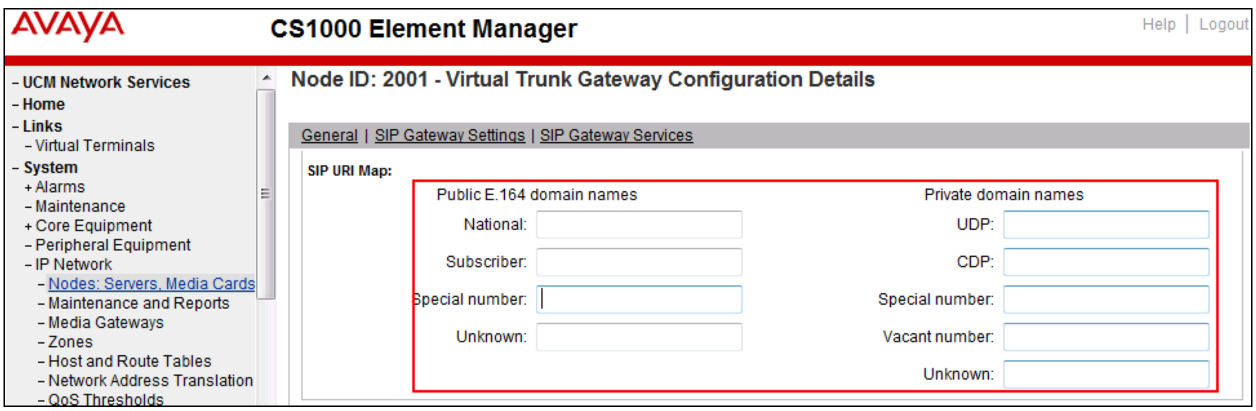
On the same page, scroll down to the **SIP URI Map** section as shown in the screenshot below. The URI Map settings were set to blank to disable the “phone-context” from being sent because it is not required by Windstream.

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

- **National**: Set the field to blank.
- **Subscriber**: Set the field to blank.
- **Special Number**: Set the field to blank.
- **Unknown**: Set the field to blank.

Under the **Private Domain Names**:

- **UDP**: Set the field to blank.
- **CDP**: Set the field to blank.
- **Special Number**: Set the field to blank.
- **Vacant number**: Set the field to blank.
- **Unknown**: Set the field to blank.



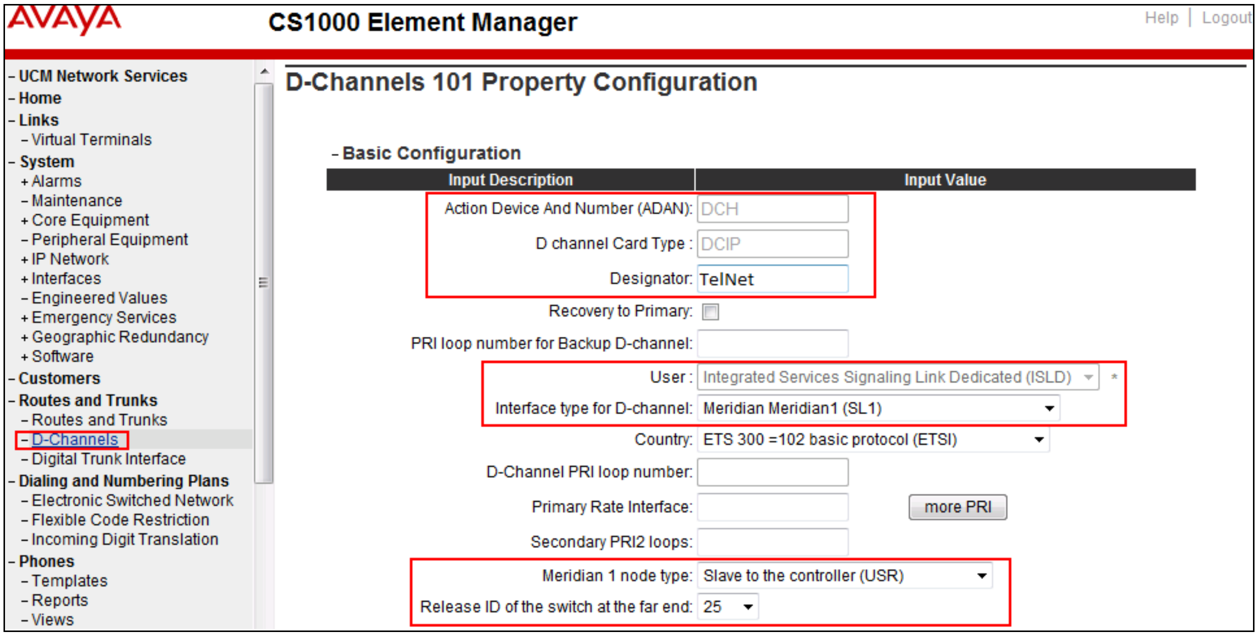
Then click **Save** button (not shown) and synchronize the new configuration (see **Section 5.2.3**).

5.5.3. Administer Virtual D-Channel

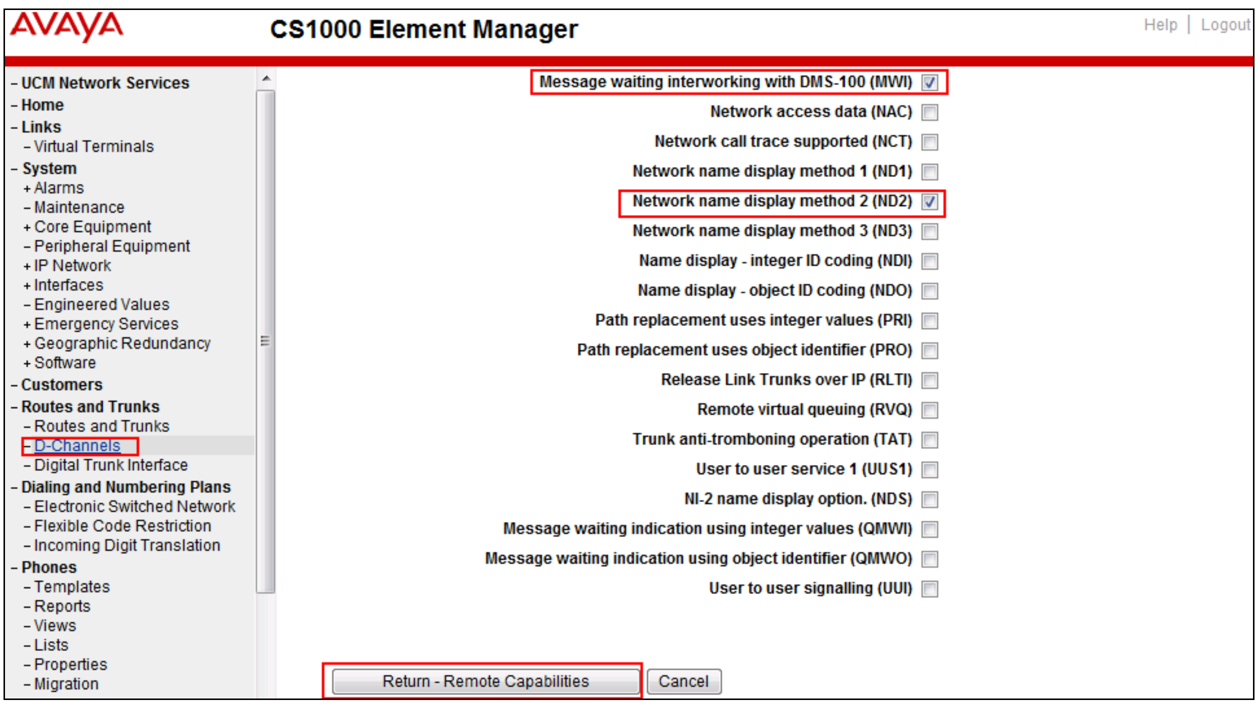
To create a D-Channel, select **Routes and Trunks → D-Channels** from the left pane to display the **D-Channels** screen (not shown). In the **Choose a D-Channel Number** field, select an available D-channel from the drop-down list (not shown). Click on **to Add** button (not shown)**.**

The **D-Channels Property Configuration** of DCH **101** is shown in the screenshot below. Enter the following values for the specified fields, and retain the default values for the remaining fields.

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



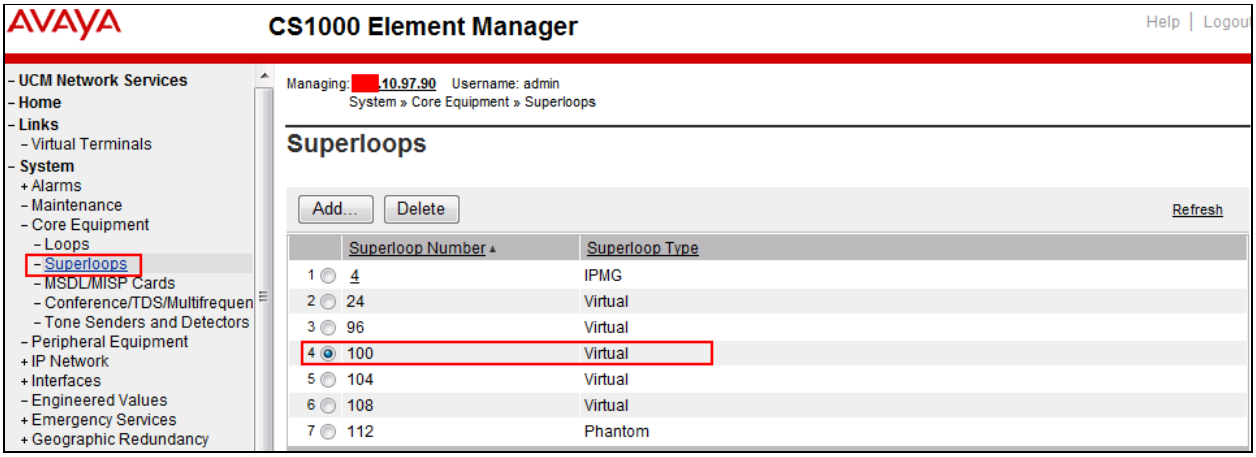
Click on the **Basic Options** then click on the **Edit** button at the **Remote Capabilities** (**RCAP**) attribute (not shown). The **Remote Capabilities Configuration** page will appear. Then check on the **Message waiting interworking with DMS-100 (MWI)** and the **Network name display method 2 (ND2)** checkboxes as shown in the screenshot below.



Click **Return – Remote Capabilities** button then click **Submit** button (not shown).

5.5.4. Administer Virtual Super-Loop

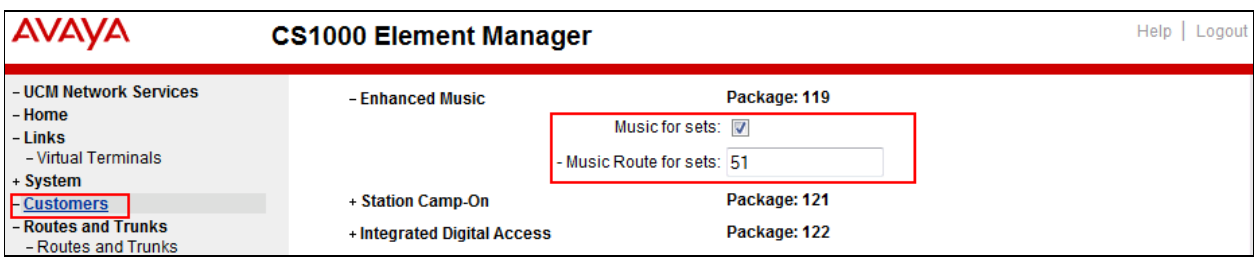
To add a virtual loop, select **System → Core Equipment → Superloops** from the left pane to display the **Superloops** screen. If the Superloop does not exist, click "**Add**" button to create a new one as shown in the screenshot below. In this example, Superloop **100** was added.



5.5.5. Enable Music for Customer Data Block

To enable music for a customer, select **Customers** in the left pane. The **Customers** screen is displayed (not shown). Click on the link associated with the appropriate customer, in this case is **01**. The **Customer 01 Edit** page will appear (not shown). Select the **Feature Packages** option from this page (not shown).

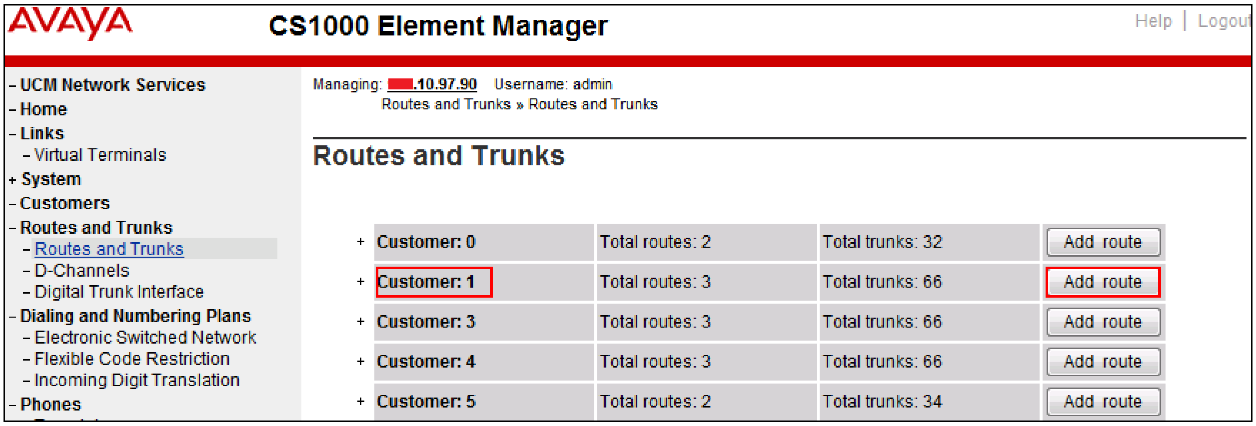
The screen is populated with a list of F**eature Packages**. Select **Enhanced Music** to edit its parameters. Check **Music for sets** to enable music for Customer **01**, define **Music Route for sets 51** as shown in the red box of screenshot below. The CS1000 has been pre-configured with music route **51**.



Scroll down to the bottom of the screen and click **Save** button (not shown).

5.5.6. Administer Virtual SIP Route

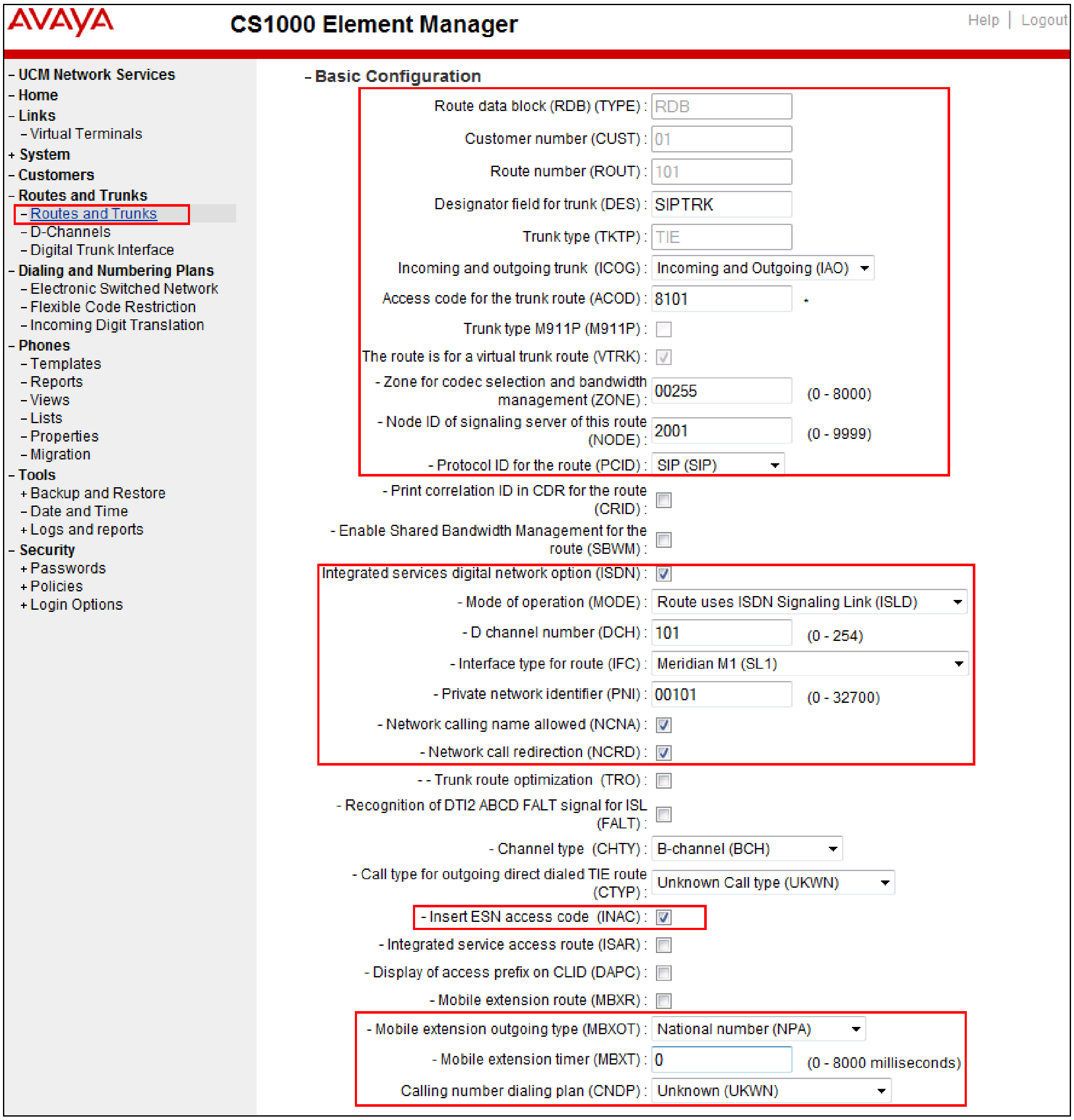
To create a SIP Route, select **Routes and Trunks → Routes and Trunks** from the left pane to display the **Routes and Trunks** screen. In this example, the new route is added under the **Customer 1**. Click **Add route** button as shown in the screenshot below.



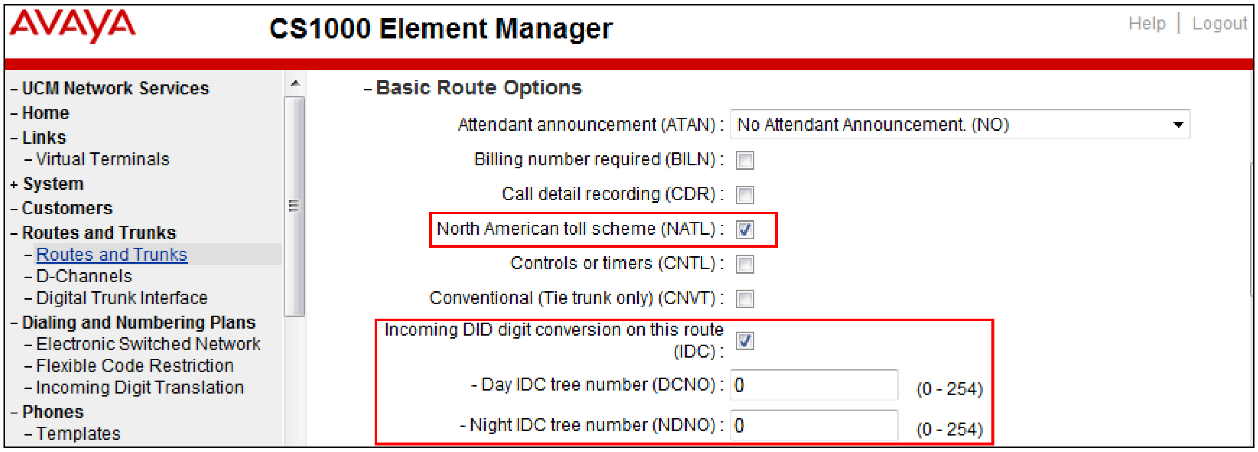
The **Customer 1**, New **Route Configuration** screen is displayed (not shown). Scroll down until the **Basic Configuration** section is displayed and enter the following values for the specified fields, and retain the default values for the remaining fields as shown in the screenshot below.

- **Route Number (ROUT)**: Select an available route number, e.g. **101**.
- **Designator field for trunk (DES)**: A descriptive text.
- **Trunk Type (TKTP)**: **TIE trunk data block (TIE)**.
- **Incoming and Outgoing trunk (ICOG)**: **Incoming and Outgoing (IAO)**.
- **Access Code for the trunk route (ACOD)**: An available access code.
- Check the field **The route is for a virtual trunk route (VTRK**), to enable additional fields to appear.
- For the **Zone for codec selection and bandwidth management (ZONE)** field, enter zone **255** (created in **Section 5.4.2**).
- For the **Node ID of signalling server of this route (NODE)** field, enter the node number **2001** (created in **Section 5.2.1**).
- Select **SIP (SIP)** from the drop-down list for **the Protocol ID for the route (PCID)** field.
- Check the **Integrated Services Digital Network option (ISDN)** checkbox to enable additional fields to appear. Enter the following values for the specified fields, and retain the default values for the remaining fields.
    - **Mode of operation (MODE)**: **Route uses ISDN Signalling Link (ISLD)**.
    - **D channel number (DCH)**: D-Channel number **101** (created in **Section 5.5.3**).
    - **Network calling name allowed (NCNA)**: Checked.
    - **Network call redirection (NCRD)**: Checked.
    - **Insert ESN access code (INAC):** Checked.
    - **Mobile extension outgoing type (MBXOT)**: Select **National number (NPA)**.

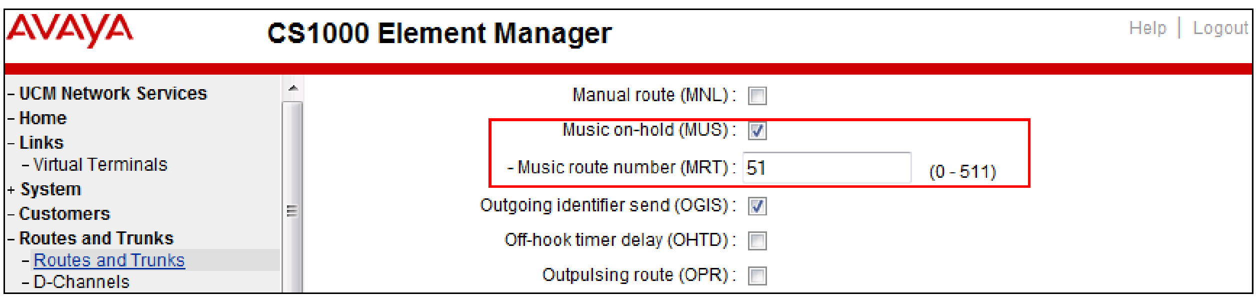
- **Mobile extension timer (MBXT)**: Define an appropriate value to meet the certain deployment at enterprise network. For this compliance test, the default value of 0 ms is used.
- **Calling number dialling plan (CNDP)**: UnKnown (UKWN).



Click on **Basic Route Options**, check **North American toll scheme (NATL)** and **Incoming DID digit conversion on this route (IDC)**.Enter a value of **0** for both **Day IDC Tree Number (DCN0)** and **Night IDC Tree Number (NDCN0)** as shown in screenshot below. The IDC is discussed in **Section 5.6.5**.



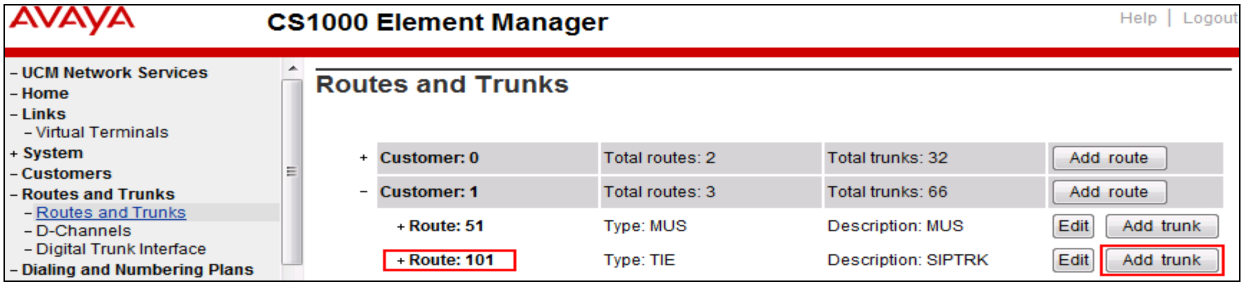
Click on **Advance Configurations** (not shown); check **Music-on-hold (MUS)** to enable music on hold on this route. Enter a value of **51** to the Music route number (MRT) box as shown in the screenshot below. The CS1000 has been pre-configured with route **51** as a music route.



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

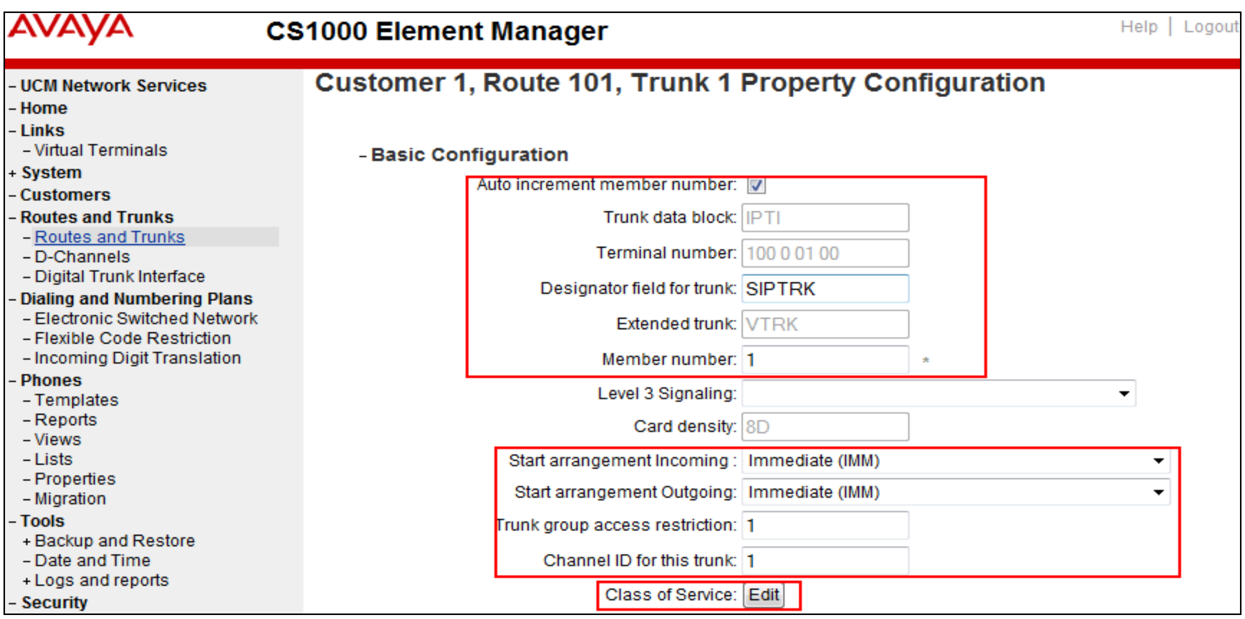
5.5.7. Administer Virtual SIP Trunks

To configure the virtual SIP Trunks, select **Route 101** that was added in **Section 5.6.6** then click click **Add trunk** button next to the newly added **Route 101** as shown in the screenshot below.

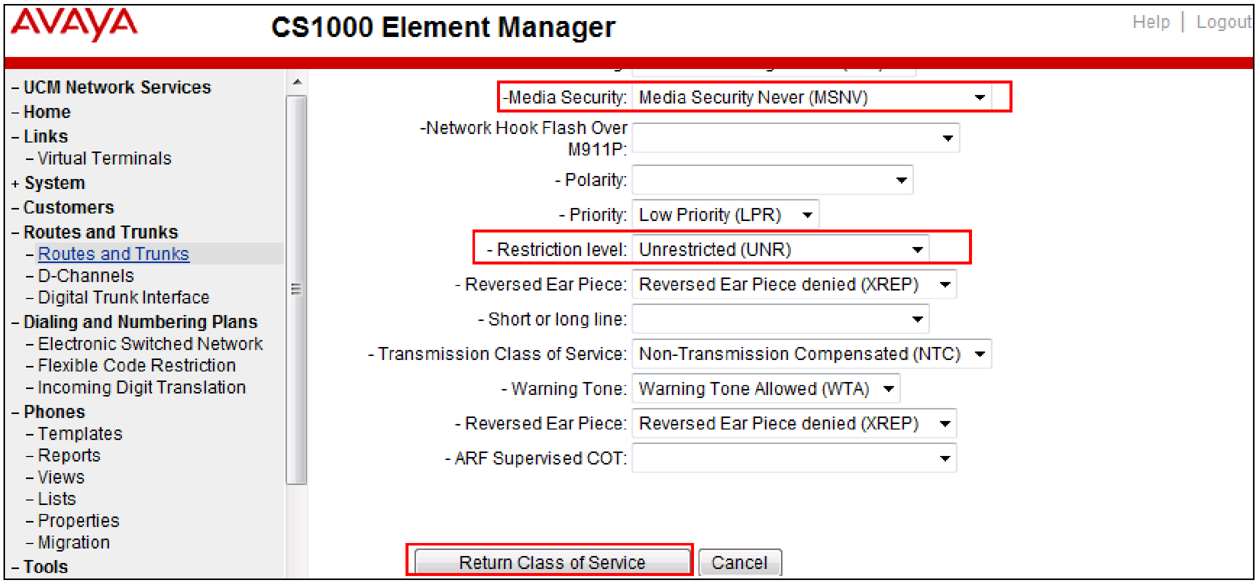


The **Customer 1, Route 101, Trunk 1 Property Configuration** is shown in the screenshot below. Enter **The Multiple trunk input number (MTINPUT)** field (not shown) to add multiple trunks in a single operation, or repeat the operation for each trunk. In the certification testing, 32 trunks were created (not shown). The following values were entered for specified fields and retain the default values for the remaining fields.

- **Trunk data block**: IP Trunk (IPTI).
- **Terminal Number**: Available terminal number, e.g. 100 0 1 0(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.
- **Start arrangement Incoming**: Immediate (IMM).
- **Start arrangement Outgoing**: Immediate (IMM).
- **Trunk Group Access Restriction**: Desired trunk group access restriction level e.g. **1**.
- **Channel ID for this trunk**: An available starting channel ID e.g. **1**.



The Media Security (SRTP) has to be disabled at the trunk level by editing the **Class of Service** (CLS) at the bottom of the basic trunk configuration page. Click **Edit** button to configure (not shown). For **Media Security**, select **Media Security Never** (**MSNV**). Select **Restriction level** as **Unrestricted (UNR)**. The remaining values are kept as default as shown in the screenshot below. Scroll down to the bottom of the screen and click **Return Class of Service** and then click **Save** button (not shown).



5.5.8. Administer Calling Line Identification Entry

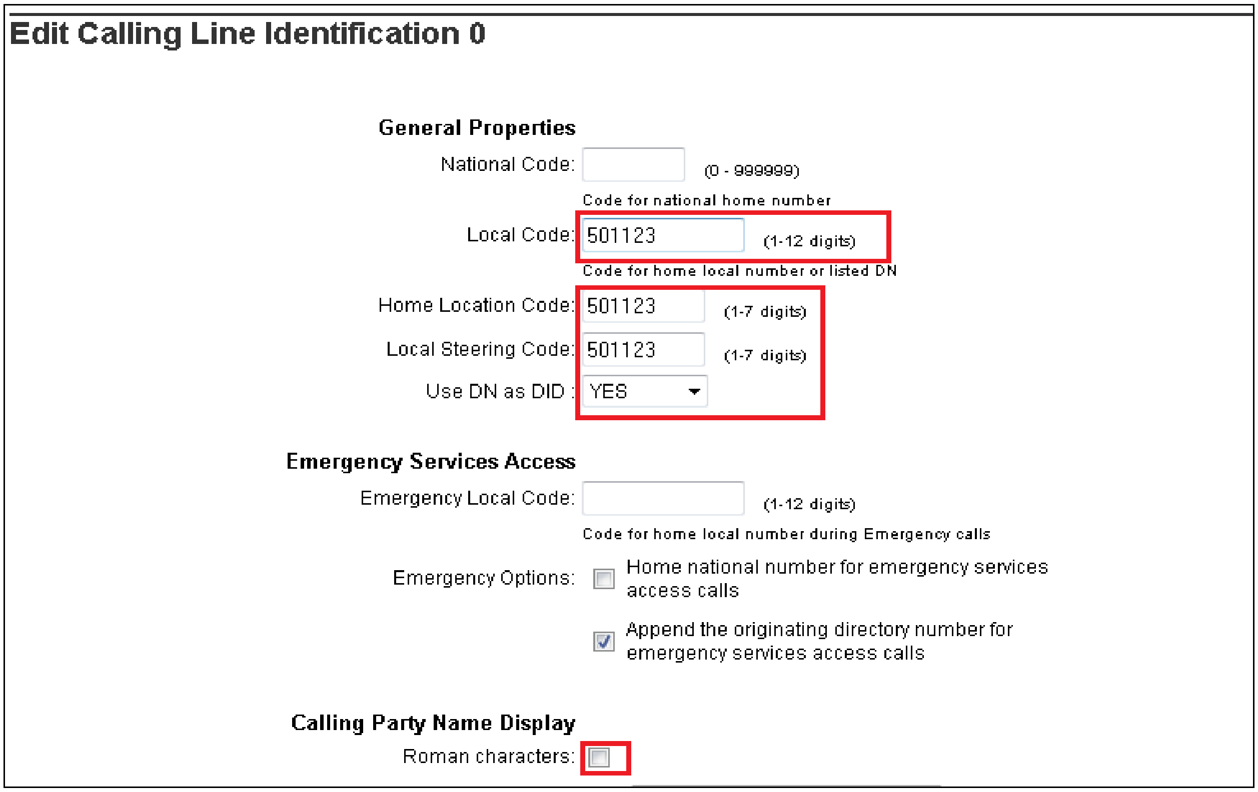
To create Calling Line Identification Entry, select **Customers → 01 → ISDN and ESN Networking**. Click **Calling Line Identification Entries** link at the bottom of the page (not shown).

On the **Calling Line Identification Entries** page (not shown), click **Add**. Add entry **0** as shown in the screenshot below.

- **National Code**: Leave as blank.
- **Local Code**: Input a prefix what was assigned by the service provider, in this case it is 6 digits **501XXX**. This **Local Code** is used for call display purpose of outgoing call configuration in **Section 5.6.6** where the Special Number is associated with Call Type = NONE. Note that for the security reason the last 3 digits is hidden by XXX but in the real deployment it should be entered with full 6 digits, e.g. 501123.
- **Home Location Code**: Input prefix that was assigned by the service provider, in this case it is 6 digits **501XXX**. This **Home Location Code** is used for call display purpose of outgoing call configuration in **Section 5.6.6** where the Special Number is associated with Call Type = National (NPA).
- **Local Steering Code**: Input a prefix that was assigned by the service provider, in this case it is 6 digits **501XXX**. This **Local Steering Code** is used for call display purpose of

outgoing call configuration in **Section 5.6.6** where the Special Number is associated with Call Type = National (NXX).

- **Use DN as DID**: Select **YES**. Note that if selecting YES in this option, local number of the CS1000 system will be added to the end of the local code above and make it up as DID numbers known by Windstream.
- **Calling Party Name Display**: Uncheck the **Roman characters** field.
- Click **Save** button (not shown).



5.5.9. Enable External Trunk to Trunk Transferring

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

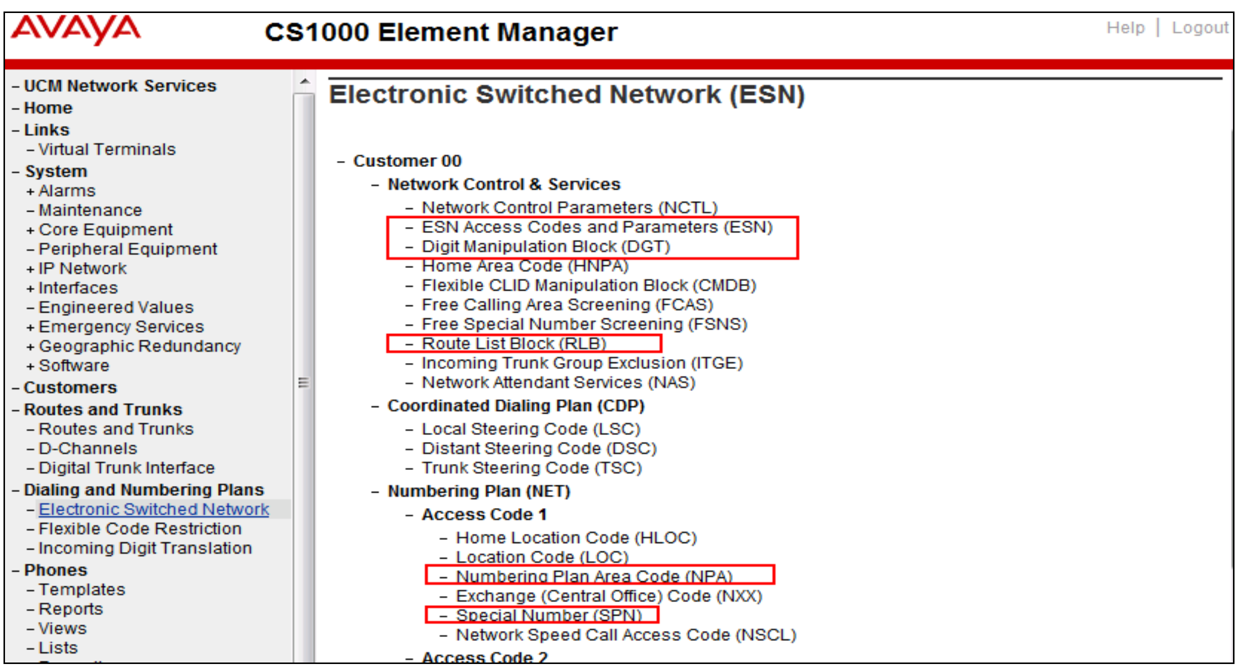
- Log into Call Server CLI (please refer to **Section 5.1.2** for more detail).
- Allow **External Trunk To Trunk Transferring** for **Customer Data Block** by using LD 15.

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

TYPE NET_DATA
CUST 1
OPT
…
TRNX YES
EXTT YES
…
```

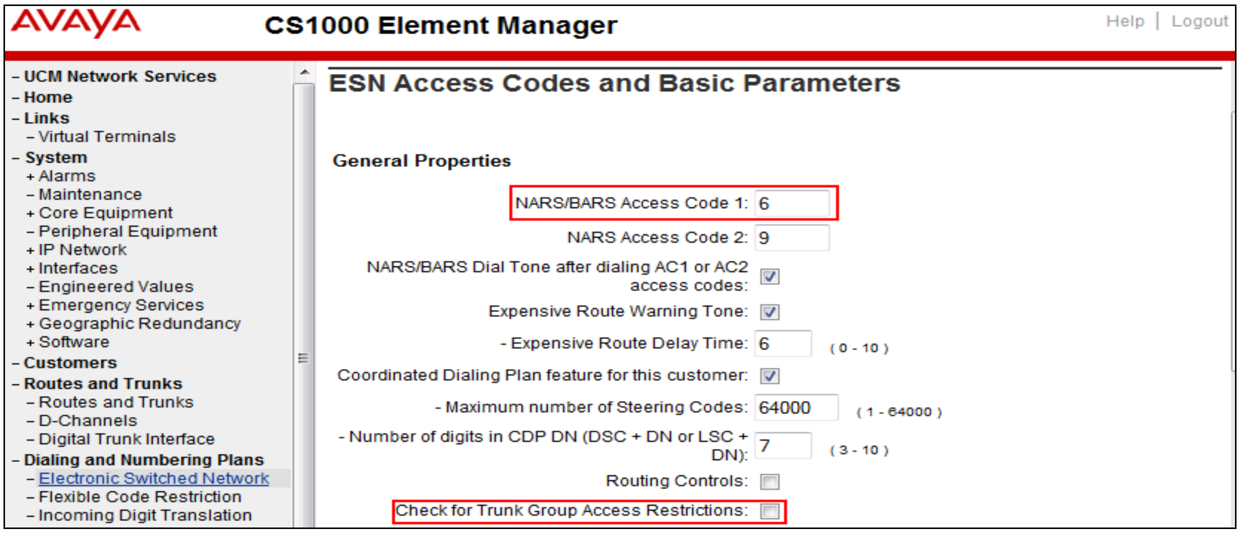
5.6. Administer Dialing Plans

5.6.1. Define ESN Access Codes and Parameters (ESN)

To configure Electronic Switched Network (ESN) parameters, select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (**ESN**) screen. Select **ESN Access Code and Parameters (ESN)** as shown in the screenshot below.



In the **ESN Access Codes and Basic Parameters** page, define **NARS/ BARS Access Code 1** and disable **Check for Trunk Group Access Restrictions** as shown in the screenshot below. Click **Submit** button (not shown).



5.6.2. Associate Numbering Plan Area Code (NPA) and Special Number (SPN) calls to ESN Access Code 1

This section shows the configuration to associate the NPA and SPN to ESN Access Code 1.

- Log into Call Server CLI (refer to **Section 5.1.2** for more detail).
- In **LD 15**, change Customer **Net_Data** block by disabling **NPA** and **SPN** to be associated to **AC2** (Access Code 2). It means Access Code 1 will be used for NPA and SPN calls.

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

TYPE NET_DATA
CUST 1
OPT
AC2 xNPA xSPN
FNP
CLID
…
```

Verify Customer Net_Data block by using **LD 21**. The **NPA** and **SPN** parameters are now moved to ESN Access Code 1 (AC1).

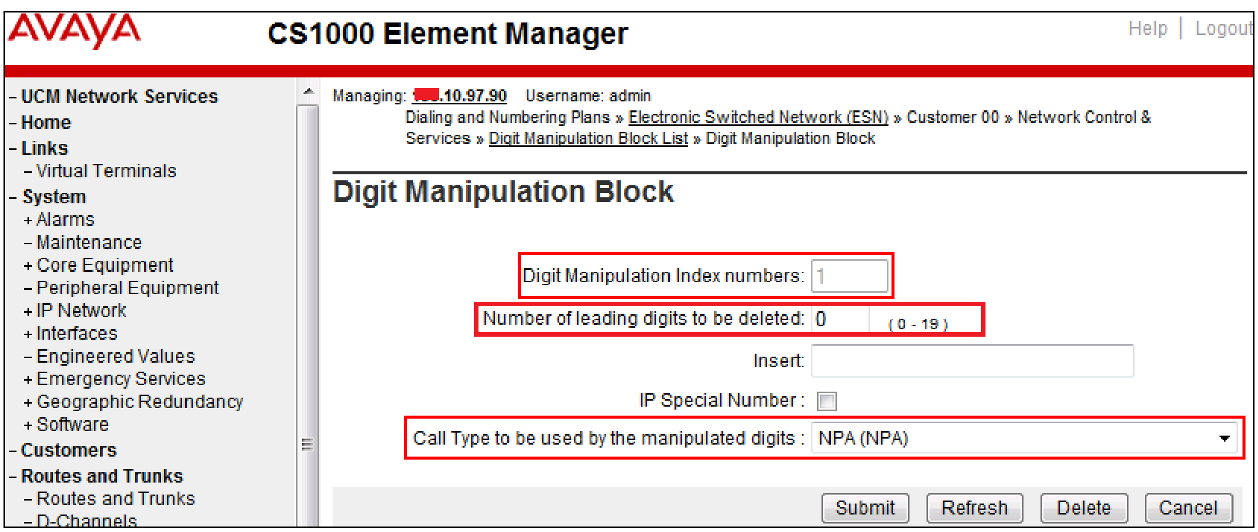
```
>ld 21
PT1000

REQ: prt
TYPE: net
TYPE NET_DATA
CUST 1

TYPE NET_DATA
CUST 01
OPT RTA
AC1 INTL NPA SPN NXX LOC
AC2
FNP YES
…
```

5.6.3. Administer Digit Manipulation Block (DMI)

To create a DMI entry, select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (**ESN**) screen (not shown). Then select **Digit Manipulation Block** (**DGT**) as shown in Section 5.6.1.

In the **Choose a DMI Number** field, select an available DMI from the drop-down list and click **to Add** (not shown). The screeshot below shows **DMI 1** is created with following values.

- **Number of leading digits to be deleted** 0.
- **Call Type to be used by the manipulated digits**:NPA (NPA).
- Click **Submit** button.



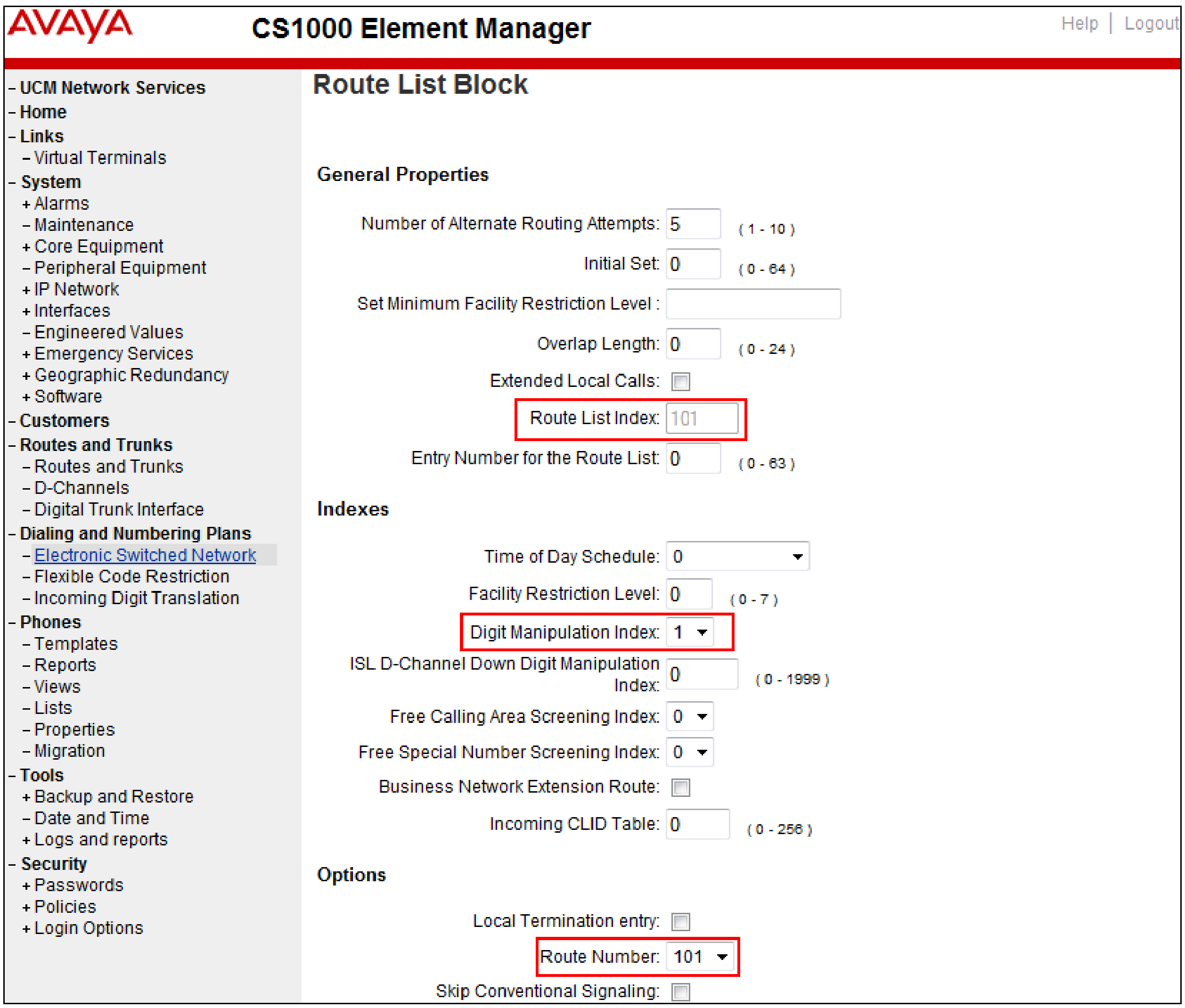
5.6.4. Administer Route List Block (RLB)

This section shows how to add a RLB associated with the DMI 1 created in **Section 0**.

To create **RLB 101** for the certification testing, select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (**ESN**) screen then select **Route List Block** (**RLB**) as shown in **Section 5.6.1**.

Select an available value, e.g. **101** in the textbox for the **route list index** and click on the "**to Add"** button (not shown). Enter the following values for the specified fields as shown in the screenshot below, and retain the default values for the remaining fields.

- **Route number (ROUT)**: **101** (created in **Section 5.5.6**).
- **Digit Manipulation Index (DMI)**: **1** (created in **Section 0**).



On the same page, scroll down to the bottom of the screen, and click **Submit** button (not shown).

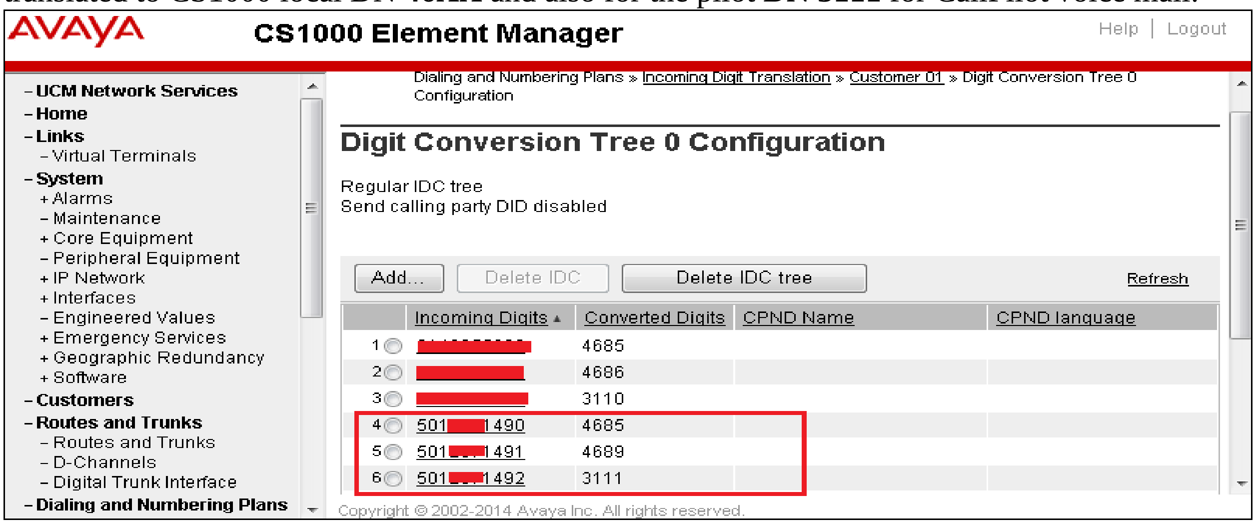
5.6.5. Administer Incoming Digit Translation (IDC)

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

To create an IDC, select **Dialing and Numbering Plans → Incoming Digit Translation** from the left pane to display the **Incoming Digit Translation** screen then click on the **Edit IDC** button (not shown).

Click on **New DCNO** to create a digit translation entry (not shown). In this example, **Digit Conversion Tree Number** (**DCN0**) **0** was created. Detailed configuration of the DCNO is shown in screenshot below. The **Incoming Digits** can be added to map to the **Converted Digits** which would be the CS1000 DN. This DCN0 has been assigned to Route 101 as shown in **Section 5.5.6**.

In the following configuration, incoming calls from PSTN with prefix **501XXX14XX** will be translated to CS1000 local DN **46XX** and also for the pilot **DN 3111** for CallPilot voice mail.



5.6.6. Administer Outbound Call - Special Number

Special Numbers are configured to be used for this testing. For example, **0** to reach service provider operator, **0+10** digits to reach service provider operator assistant, **011** prefix for international call, **1** for national long distance call, **411** for directory assistant and so on.

To create a Special Number, select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (**ESN**) screen (not shown). Then select **Special Number** (**SPN**) (not shown).

Enter the SPN value and then click on the "**to Add**" button (not shown). The screenshot below shows all the Special Numbers used for this testing.

Special Number: **0**

- **Flexible length**: 0 (flexible, unlimited and accept the character # to ending dial number).
- **Type of call this defined by the special number**: NONE.

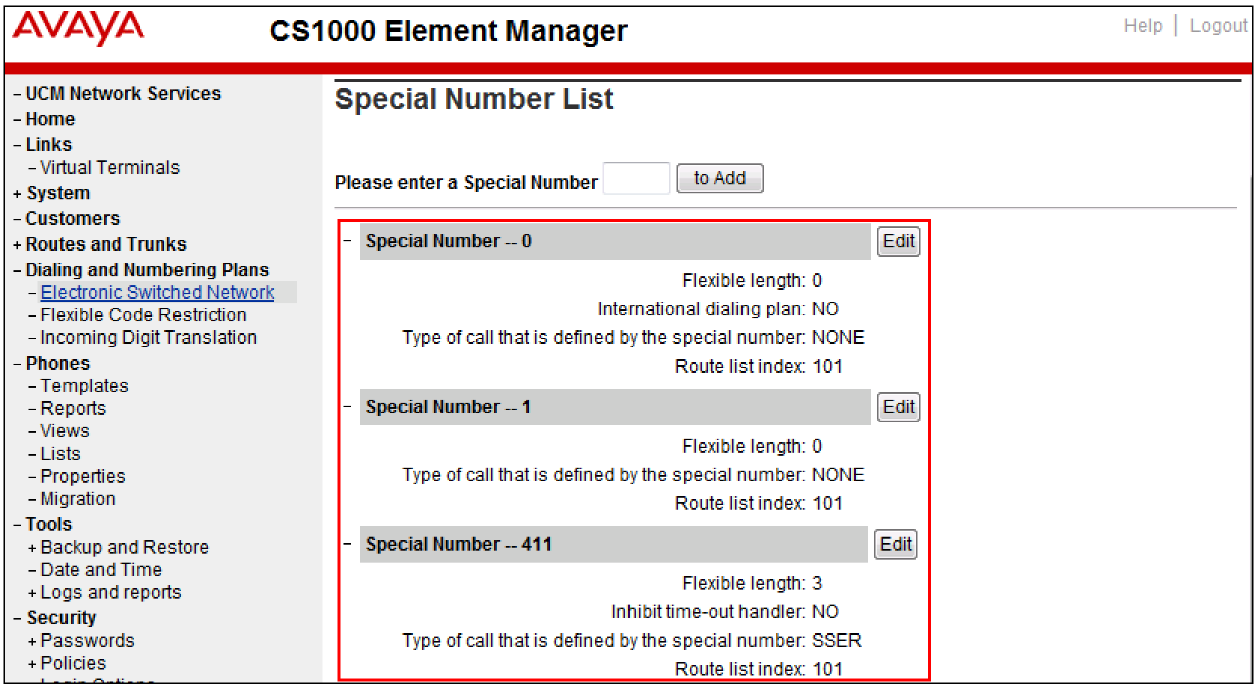
- **Route list index**: **101**, created in **Section 5.6.4**.

Special Number: **1**

- **Flexible length**: 0 (flexible, unlimited and accept the character # to ending dial number).
- **Type of call this defined by the special number**: NONE.
- **Route list index**: **101**, created in **Section 5.6.4**.

Special Number: **411**

- **Flexible length**: 3.
- **Type of call this defined by the special number**: SSER.
- **Route list index**: **101**, created in **Section 5.6.4**.

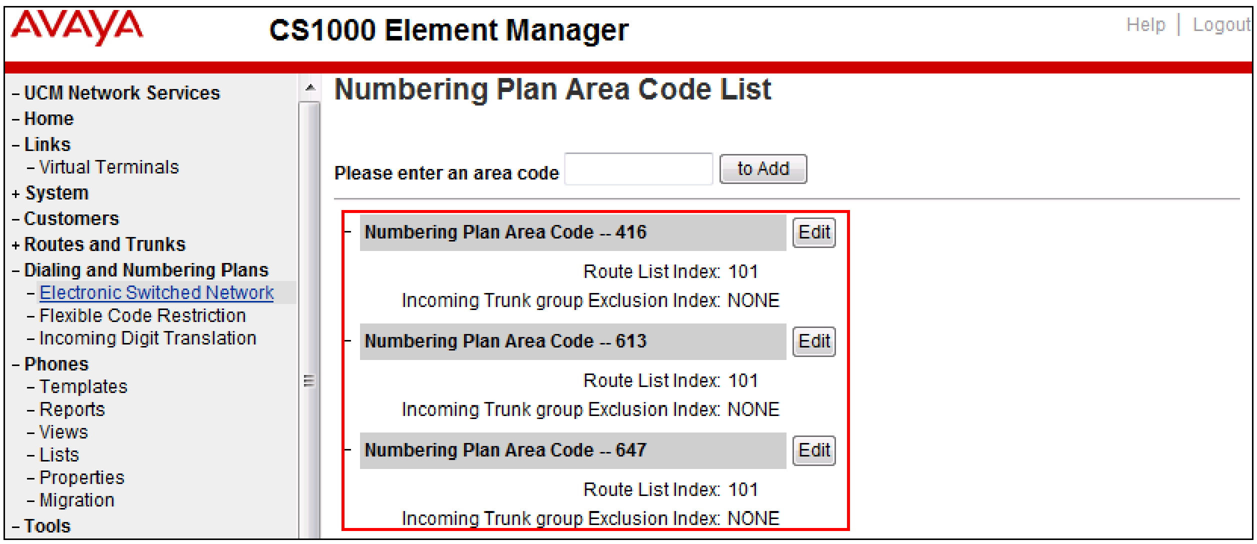


5.6.7. Administer Outbound Call - Numbering Plan Area (NPA)

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

To create a NPA number, 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 Section 5.6.1.

Enter area code desired in the textbox and click "**to Add"** button. The screenshot below shows NPA numbers **416**, **613**, and **647** were configured for this testing. These NPA numbers are associated to the SIP Trunk for 10-digit outgoing local calls.



6. Configure Avaya Aura® Session Manager

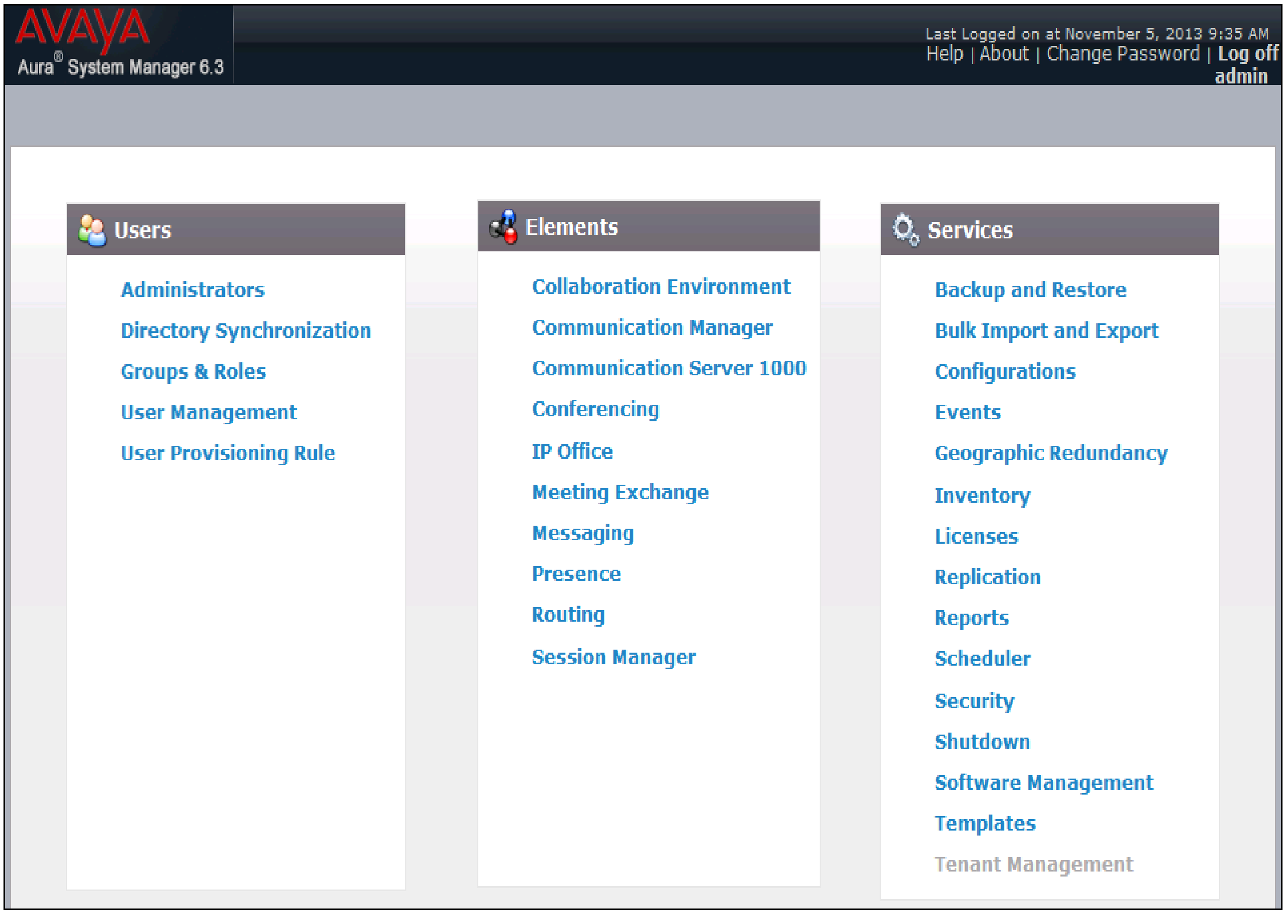
This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain.
- Logical/physical Location that can be occupied by SIP Entities.
- SIP Entities corresponding to the CS1000, Session Manager and the Avaya SBCE.
- Entity Links, which define the SIP Trunk parameters used by Session Manager when routing calls to/from SIP Entities.
- Routing Policies, which control call routing between the SIP Entities.
- Dial Patterns, which govern to which SIP Entity a call is routed.
- Session Manager, corresponding to the Session Manager server to be managed by System Manager.

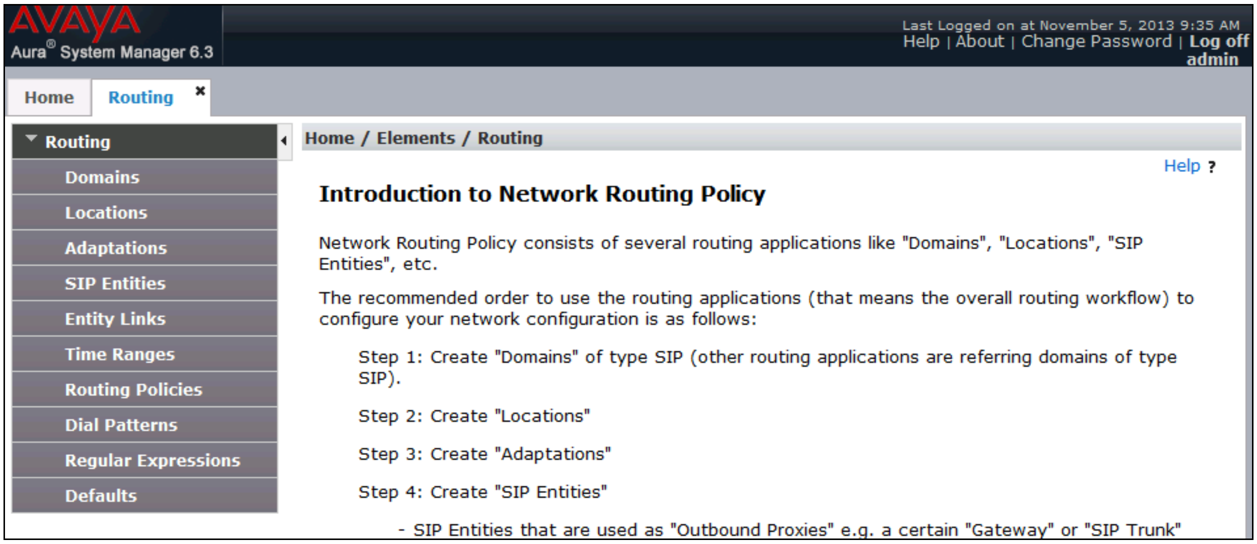
It may not be necessary to create all the items above when creating a connection to the service provider since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP domains, Location, SIP entities, and Session Manager itself. However, each item should be reviewed to verify the configuration.

6.1. System Manager Login and Navigation

Session Manager configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL "**https://<ip-address>/SMGR**", where "**<ip-address>**" is the IP address of System Manager. At the **System Manager Log On** screen, provide the appropriate credentials and click on **Login** (not shown). The initial screen shown below is then displayed.



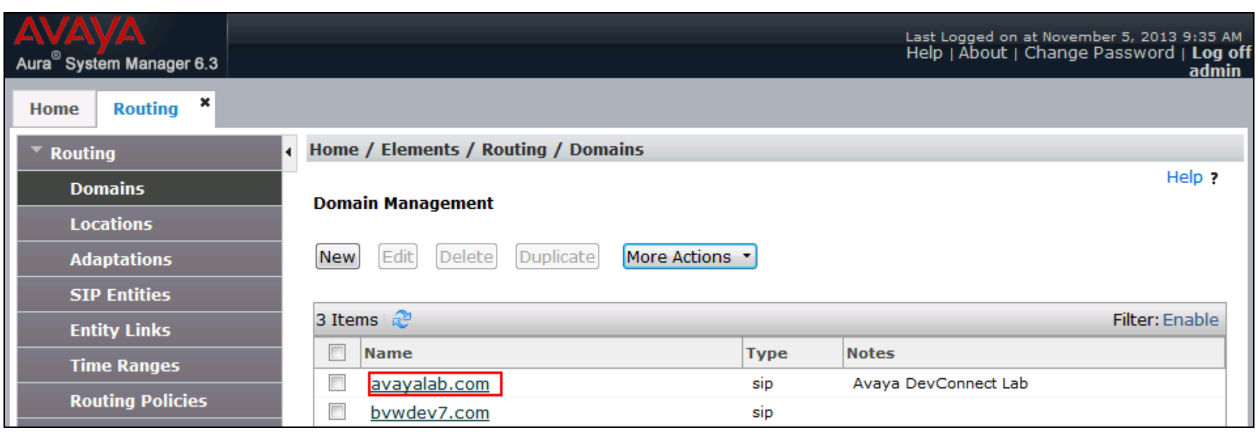
Most of the configuration items are performed in the Routing Element. Click on **Routing** in the **Elements** column to bring up the **Introduction to Network Routing Policy** screen. The navigation tree displayed in the left pane will be referenced in subsequent sections to navigate to items requiring configuration.



6.2. Specify SIP Domain

To view or change SIP domains, select **Routing → Domains**, click on the checkbox next to the name of the SIP domain and **Edit** to edit an existing domain, or the **New** button to add a domain. Click the **Commit** button (not shown) after changes are completed.

The following screen shows the list of configured SIP domains. The domain **avayalab.com** is an enterprise private SIP domain, which is defined to route incoming calls to the CS1000. Incoming calls were received with service provider public IP address **64.xx.xxx.187** which will be translated by the Avaya SBCE to **avayalab.com** to route to Session Manager. The enterprise SIP domain **avayalab.com** will be translated by the Avaya SBCE to **64.xx.xxx.187** to route to Windstream networks.



6.3. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management and call admission control. To add a location, navigate to **Routing →Locations** in the left-hand navigation pane and click the **New** button in the right pane (not shown).

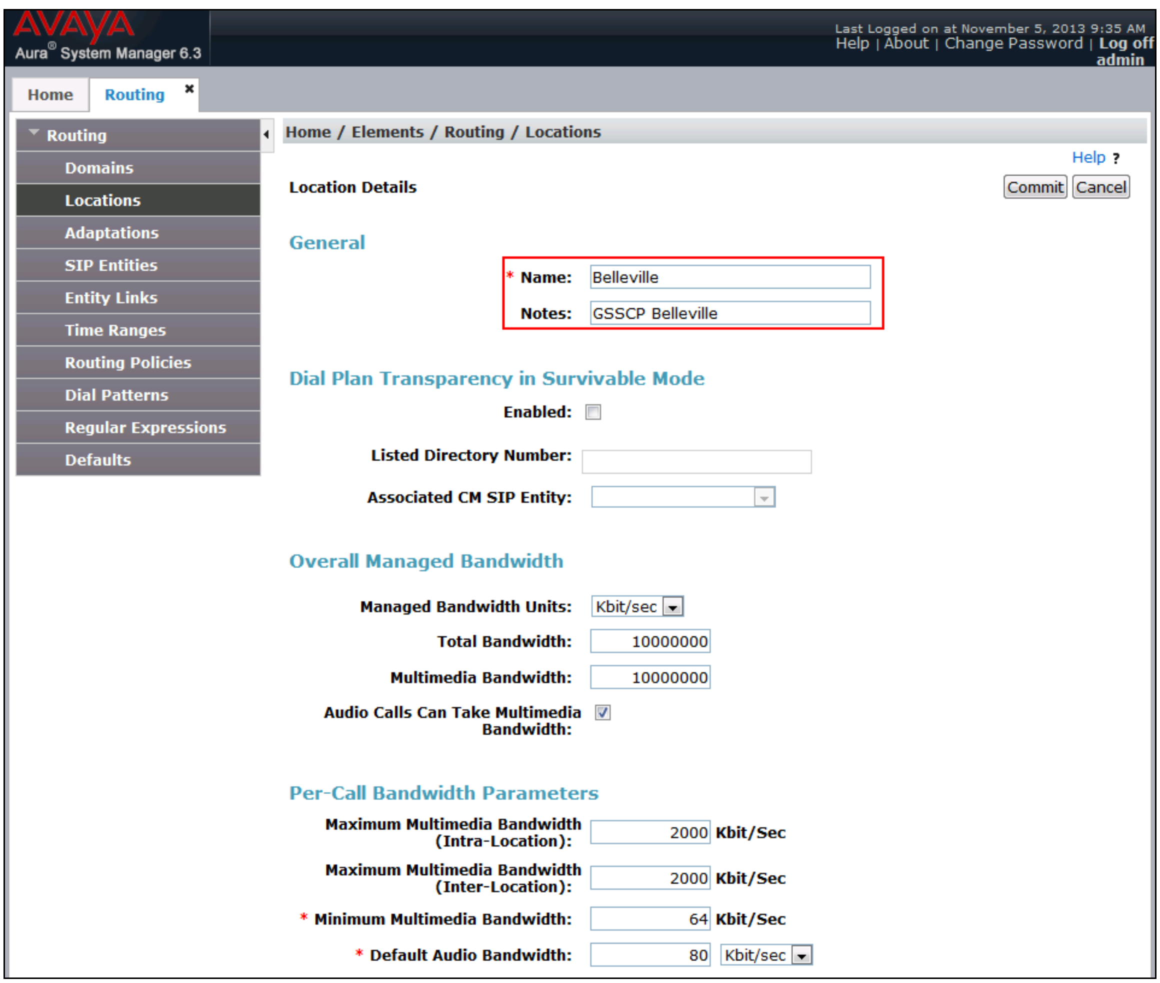
In the **General** section, enter the following values:

- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description (optional).

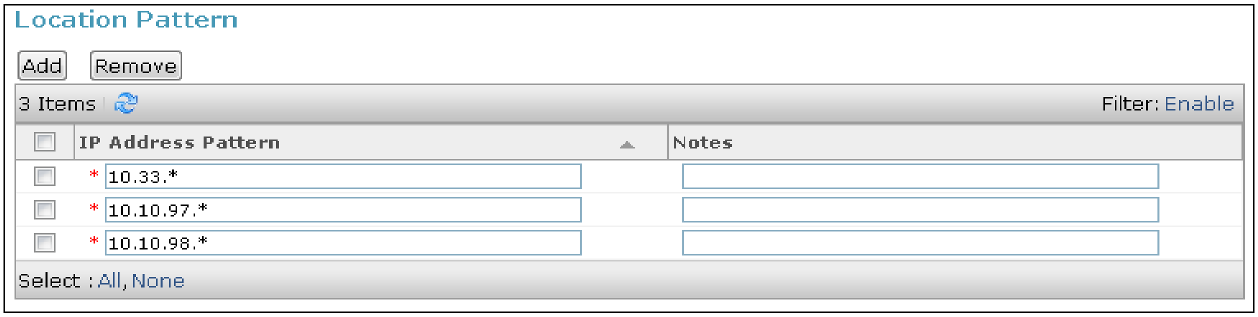
In the **Location Pattern** section, click **Add** and enter the following values:

- **IP Address Pattern:** An IP address pattern used to identify the location.
- **Notes:** Add a brief description (optional).

Displayed below are the screenshots for location **Belleville** which includes all equipment on the **10.10.97.***, **10.10.98.*** and **10.33.*** subnets including the CS1000, Session Manager, the Avaya SBCE and IP phones. Click **Commit** to save.



Continued to the screenshot above, the **Location Pattern** section is displayed as the screen below.



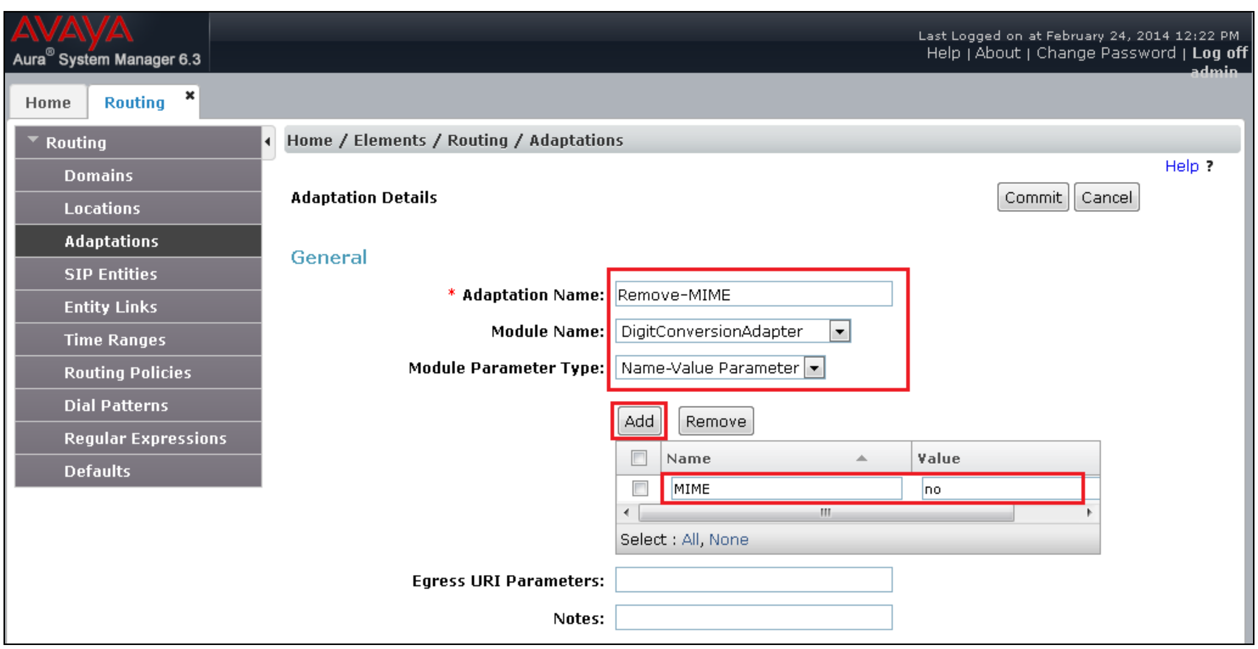
6.4. Add Adaptations

An adaptation to remove MIME multipart in INVITE message sent from the CS1000 will be created. This adaptation will be applied to SIP INVITES sent to the Avaya SBCE in next section. Navigate to **Routing → Adaptation** in the left navigation pane and click on the **New** button in the right pane (not shown).

In the General section, enter the following values. Use default values for all remaining fields:

- **Adaptation Name**: enter a descriptive name.
- **Module Name**: select **DigitConversionAdapter** in the dropdown list.
- **Module Parameter Type**: select **Name-Value Parameter** in the dropdown list.
- Click **Add** button and add a value, enter "**MIME**" in the Name field and set its value to "**no**" in the Value field as the screen shown below.

Click **Commit** button to save.



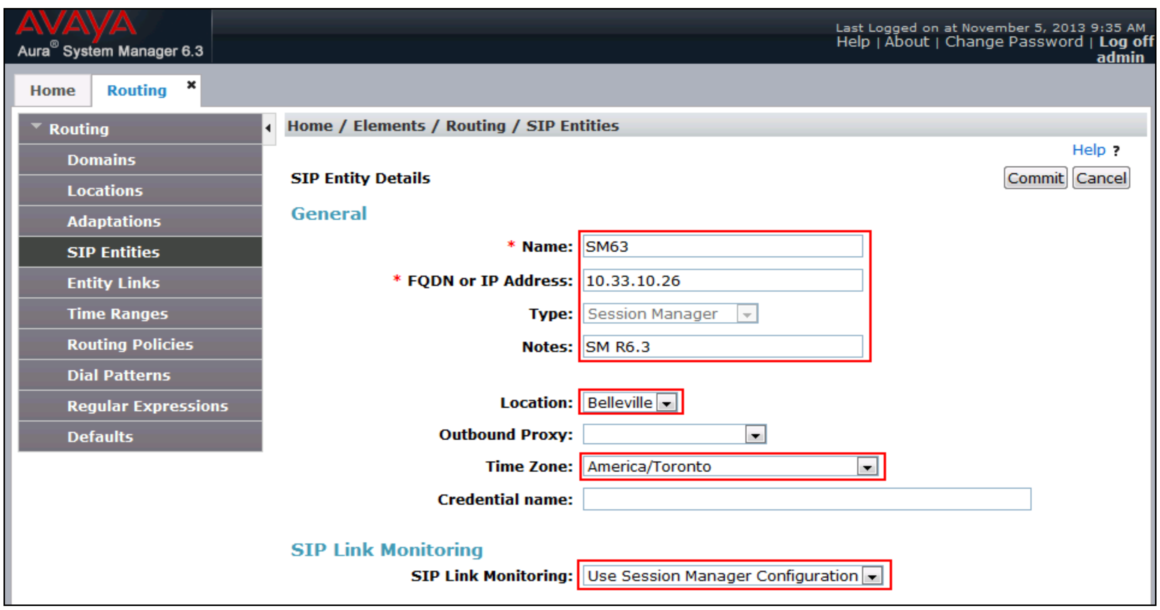
6.5. Add SIP Entities

A SIP Entity must be added for Session Manager and for each SIP telephony system connected to it which includes the CS1000 and the Avaya SBCE. Navigate to **Routing → SIP Entities** in the left navigation pane and click on the **New** button in the right pane (not shown).

In the **General** section, enter the following values. Use default values for all remaining fields:

- **Name:** Enter a descriptive name.
- **FQDN or IP Address:** Enter the FQDN or IP address of the SIP Entity that is used for SIP signaling.
- **Type:** Select **Session Manager** for Session Manager and **Other** for the CS1000 and the Avaya SBCE.
- **Location:** Select the Location defined previously.
- **Time Zone:** Select the time zone for the Location above.

The following screen shows the addition of SIP Entity for Session Manager. The IP address of Session Manager signaling interface is entered for **FQDN or IP Address**. The **SIP Link Monitoring** is kept as default **Use Session Manager Configuration**.



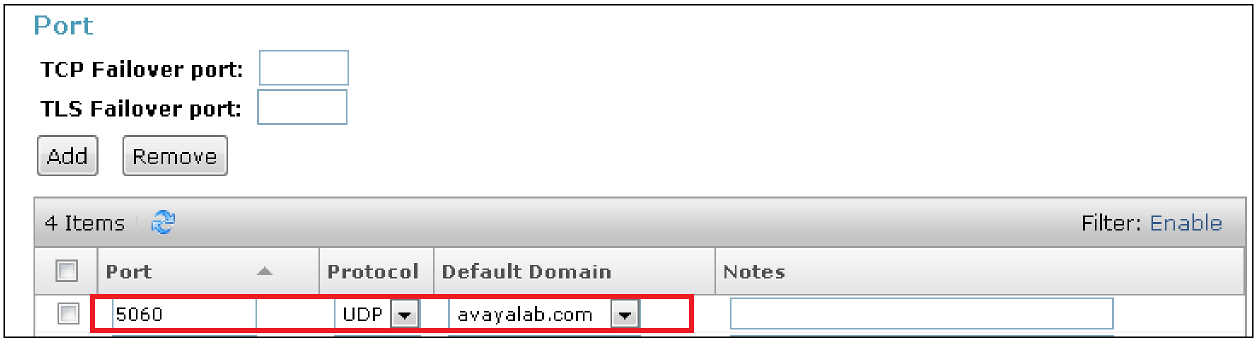
To define the ports used by Session Manager, scroll down to the **Port** section of the **SIP Entity Details** screen. This section is only present for the **Session Manager** SIP Entity.

In the **Port** section, click **Add** and enter the following values. Use default values for all remaining fields:

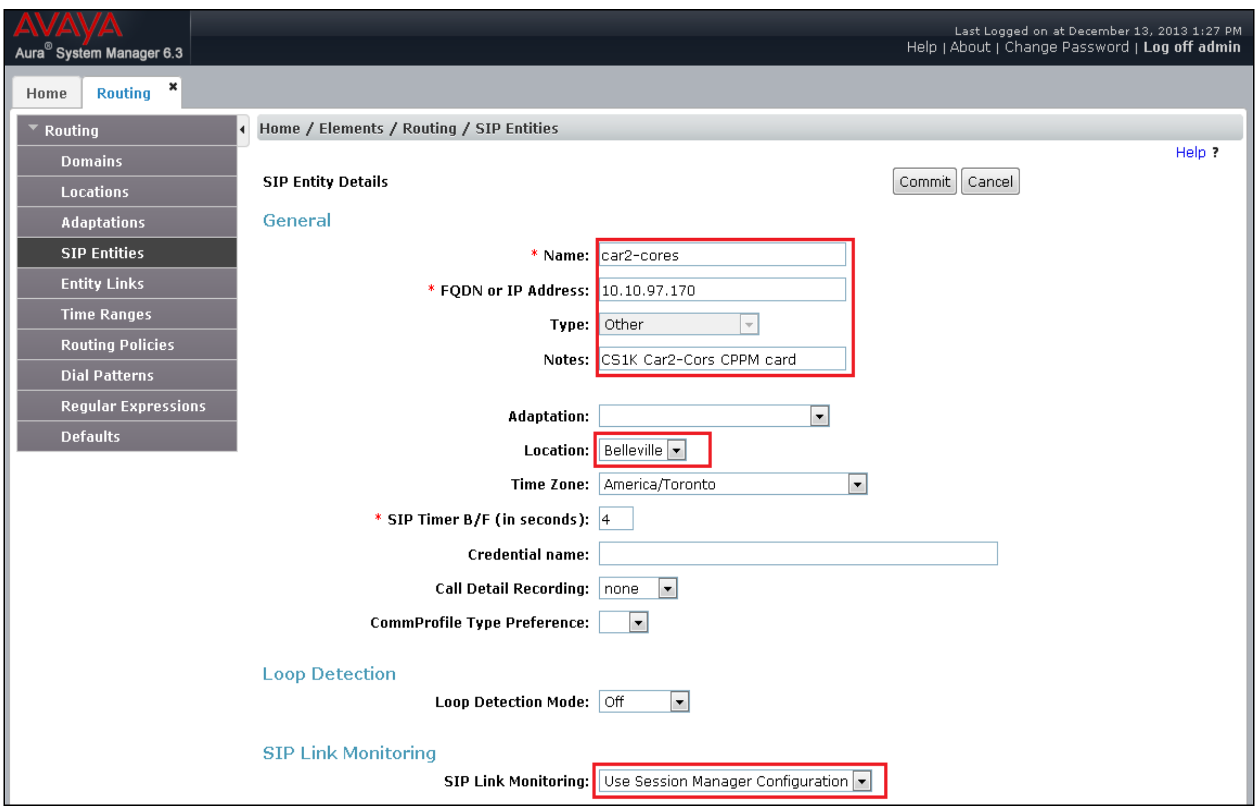
- **Port**: Port number on which the Session Manager can listen for SIP requests.
- **Protocol**: Transport protocol to be used to send SIP requests.
- **Default Domain**: The domain used for the enterprise.

Defaults can be used for the remaining fields. Click **Commit** to save.

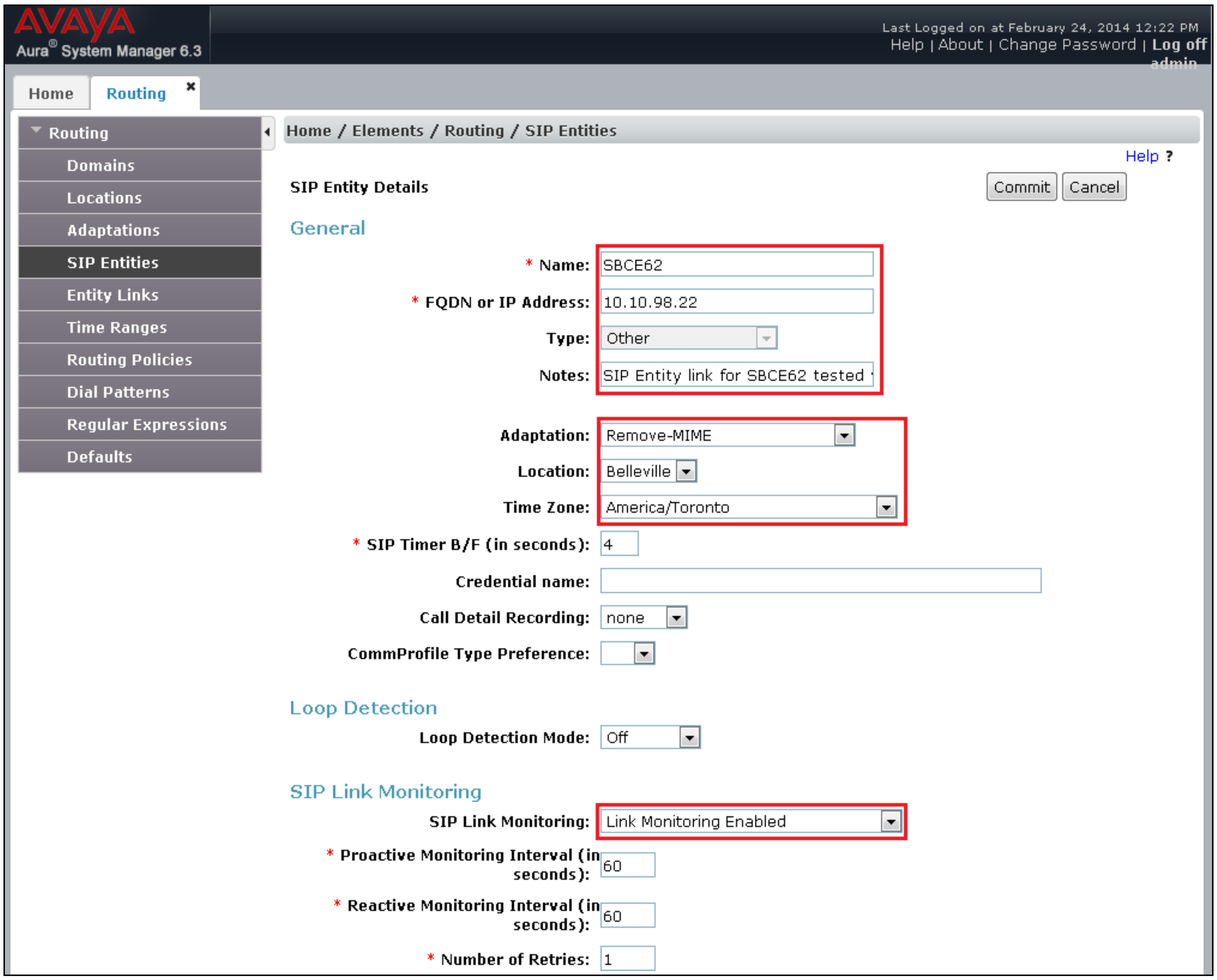
The compliance testing used **Port** entry **UDP/5060** connecting to the CS1000 for the internal enterprise calls. The **Port** entry **UDP/5060** is connecting to the Avaya SBCE for the external PSTN calls.



The following section shows the addition of SIP Entity **car2-cores** for the CS1000. The **FQDN or IP Address** field is set to the IP address of the CS1000 as **10.10.97.170**. Select **Type** is **Other**. In the compliance testing, a single SIP Entity was created for both incoming and outgoing calls consistent with the SIP Trunk created on the CS1000 in **Section 5.5**. The **SIP Link Monitoring** was set to **Use Session Manager Configuration** as default. This setting allows Session Manager to periodically send OPTIONS heartbeat to check for the status of the SIP Trunk.



The following screens show the addition of SIP Entities **SBCE62** for the Avaya SBCE. The **FQDN or IP Address** field is set to the IP address of the private network interfaces as **10.10.98.22**. The **Adaptation** is set to **Remove-MIME** as configured in the section above, and the **SIP Link Monitoring** is set to **Link Monitoring Enabled**.



6.6. Add Entity Links

A SIP Trunk between Session Manager and a telephony system is described by an Entity Link. One Entity Link was created for internal enterprise traffic between Session Manager and the CS1000. Session Manager will also have one Entity Link to the Avaya SBCE for external service provider traffic.

To add an Entity Link, navigate to **Routing → Entity Links** in the left navigation pane and click on the **New** button in the right pane (not shown). Fill in the following fields in the new row that is displayed:

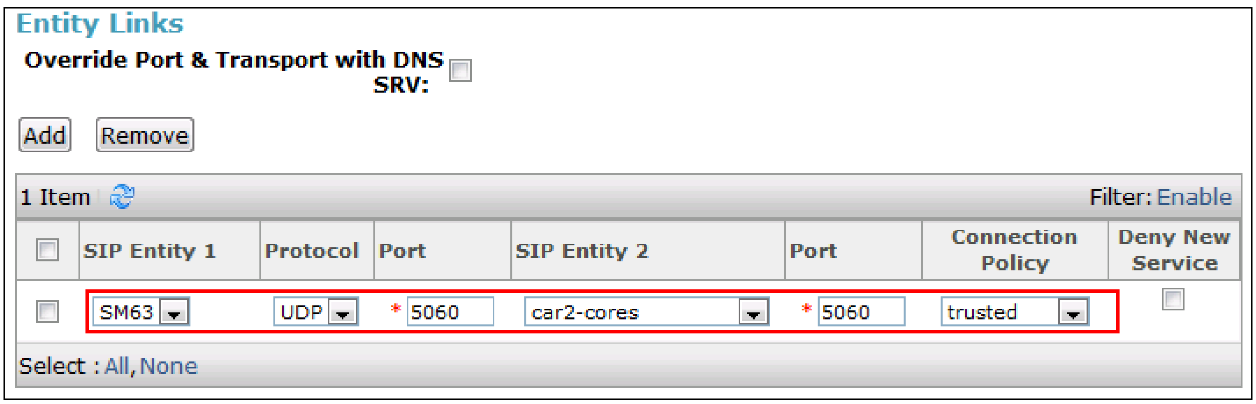
- **Name**: Enter a descriptive name.
- **SIP Entity 1**: Select the Session Manager.
- **Protocol**: Select the transport protocol used for this link.

- **Port**: Port number on which Session Manager will receive SIP requests from the far-end. For the CS1000, this must match the SIP Trunk configuration in **Section 5.5**.
- **SIP Entity 2**: Select the name of the other system. For the CS1000, select the SIP Entity **CS1000** defined in **Section 6.55**. For the Avaya SBCE, select the SIP Entity **SBCE62** defined in **Section 6.55**.
- **Port**: Port number on which the other system receives SIP requests from Session Manager. For the CS1000, this must match the SIP Trunk configuration in **Section 5.5**.
- **Connection Policy**: Select **Trusted**. **Note**: If **Trusted** is not selected, all calls from the associated SIP Entity specified in **Section 6.55** will be requested to process authentication.

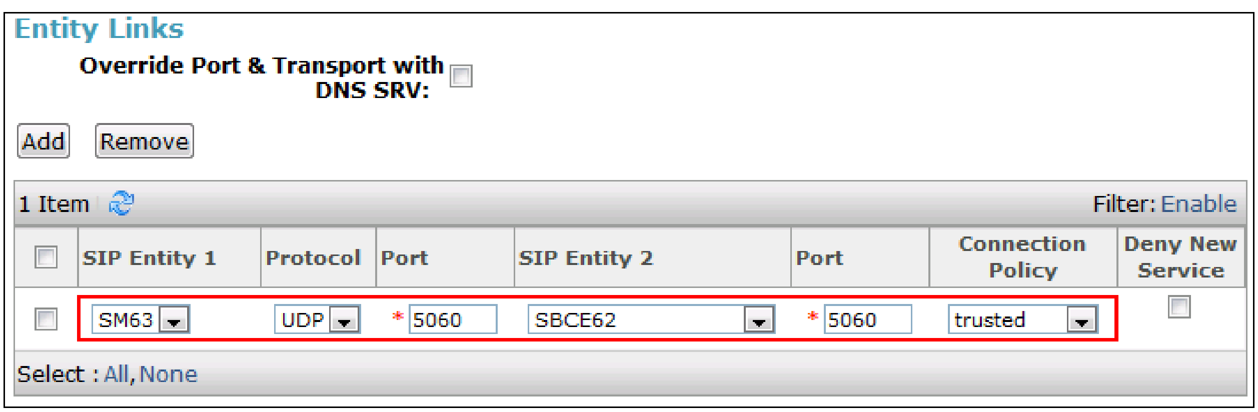
Click **Commit** to save (not shown).

The following screenshots illustrate the Entity Links between Session Manager and either CS1000 or Avaya SBCE.

Entity Link between Session Manager and the CS1000 for enterprise calls on **Port** entry use **UDP/5060**:



Entity Link between Session Manager and the Avaya SBCE for service provider calls on **Port** entry use **UDP/5060**:



6.7. Add Routing Policies

Routing Policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.55**. A separate Routing Policy was added to route incoming calls to the CS1000 and outgoing calls to the Avaya SBCE.

To add a Routing Policy, navigate to **Routing → Routing Policies** in the left navigation pane and click on the **New** button in the right pane (not shown). The following screen is displayed then fills in the following:

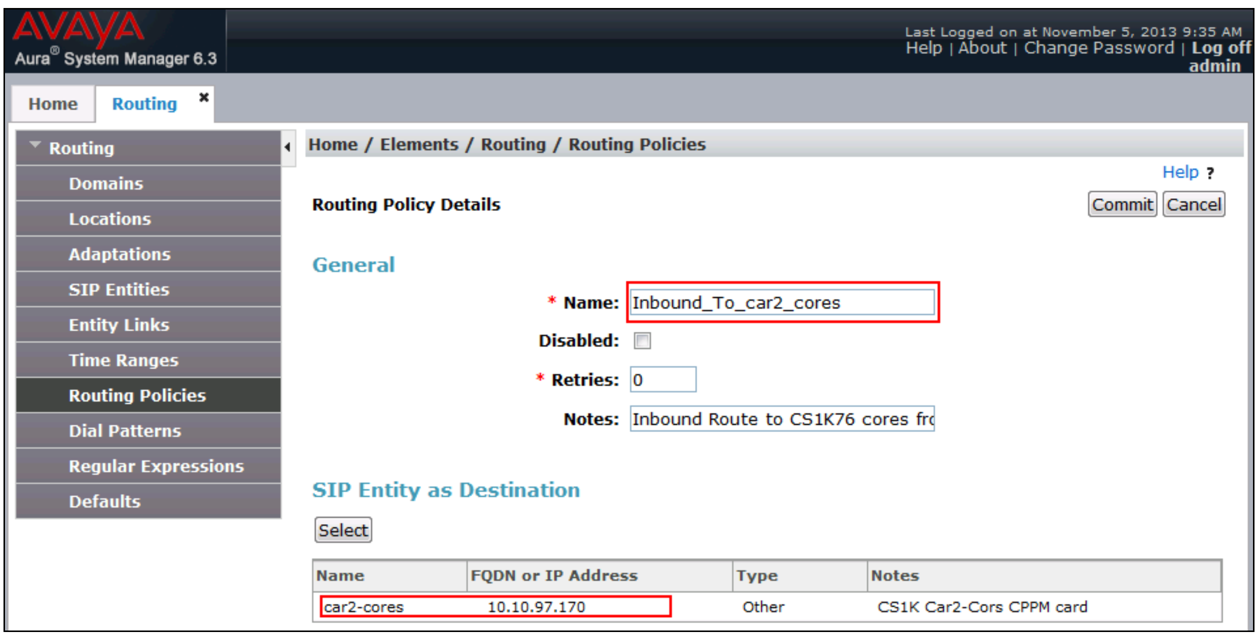
In the **General** section, enter the following values:

- **Name**: Enter a descriptive name.
- **Notes**: Add a brief description (optional).

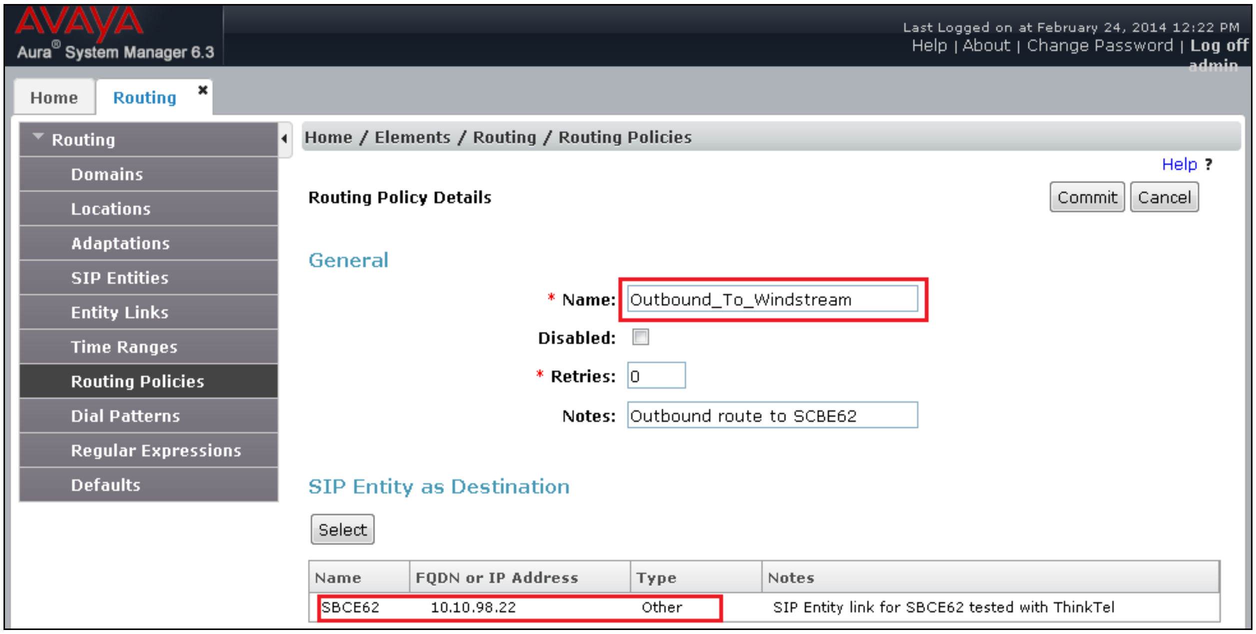
In the **SIP Entity as Destination** section, click **Select.** The **SIP Entity List** page opens (not shown). Select the appropriate SIP Entity to which this routing policy applies and click **Select**. The selected SIP Entity displays on the **Routing Policy Details** page as shown below. Use default values for remaining fields. Click **Commit** to save.

The following screens show the Routing Policies used in the compliance testing.

Routing Policy **Inbound_To_car2-cores** for incoming calls to the CS1000:



Routing Policy **Outbound_To_Windstream** for outgoing calls to the Avaya SBCE:



6.8. Add Dial Patterns

Dial Patterns are needed to route specific calls through Session Manager. For the compliance testing, Dial Patterns were needed to route calls from the CS1000 to Windstream and vice versa. Dial Patterns define which Routing Policy will be selected for a particular call based on the dialed digits, destination domain and originating location. To add a Dial Pattern, navigate to **Routing → Dial Patterns** in the left navigation pane and click on the **New** button in the right pane (not shown). Fill in the following, as shown in the screens below:

In the **General** section, enter the following values:

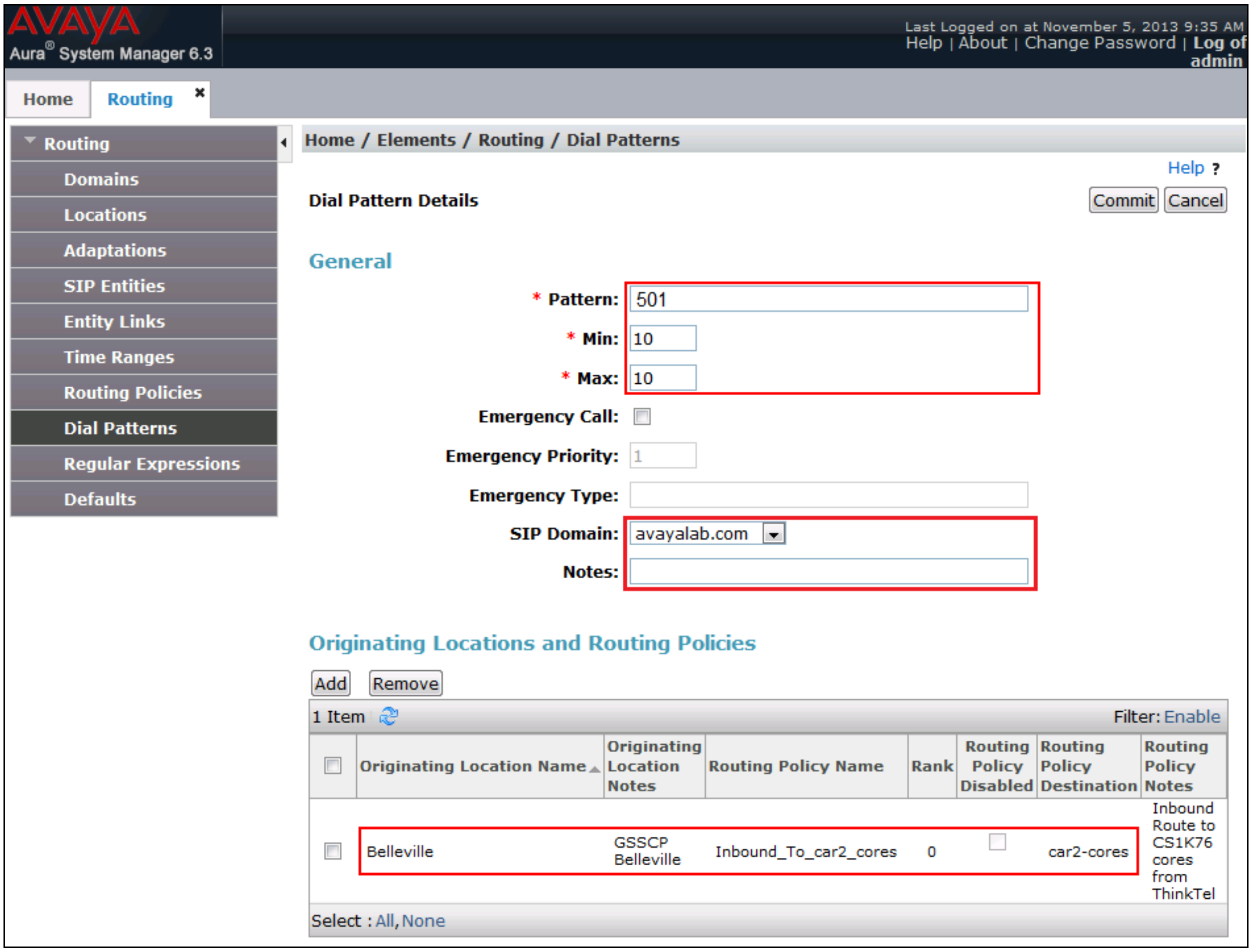
- **Pattern**: Enter a dial string that will be matched against the "Request-URI" of the call.
- **Min**: Enter a minimum length used in the match criteria.
- **Max**: Enter a maximum length used in the match criteria.
- **SIP Domain**: Enter the destination domain used in the match criteria.
- **Notes**: Add a brief description (optional).

In the **Originating Locations and Routing Policies** section, click **Add**. From the **Originating Locations and Routing Policy List** that appears (not shown), select the appropriate Originating Location for use in the match criteria. Lastly, select the Routing Policy from the list that will be used to route all calls that match the specified criteria. Click **Select**.

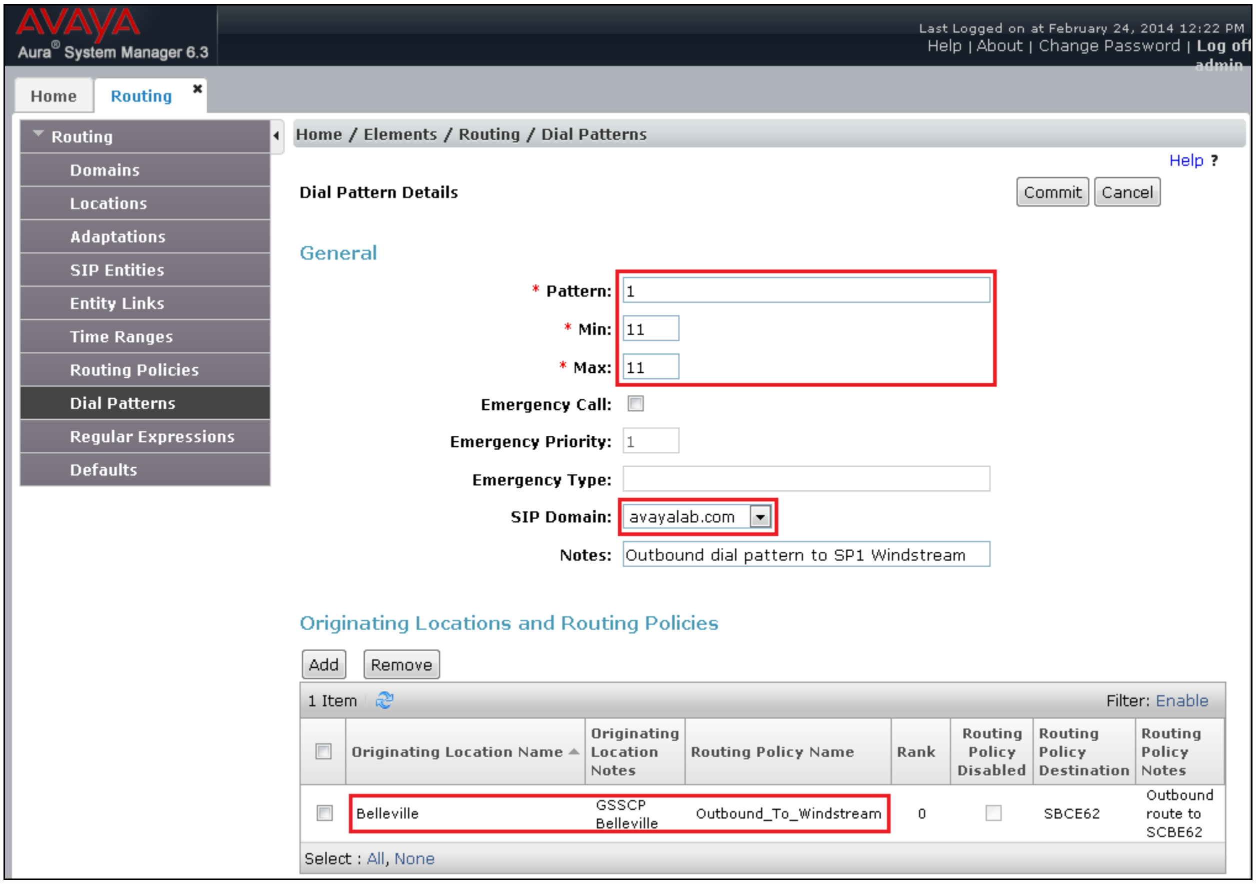
Default values can be used for the remaining fields. Click **Commit** to save.

Two examples of the dial patterns used for the compliance testing are shown below, one for outgoing calls from the enterprise to the PSTN and one for incoming calls from the PSTN to the enterprise. Other outgoing dial patterns e.g. **011** international calls, **411** directory assistance calls, etc., were similarly defined.

The first example shows a Dial Pattern for incoming calls that 10-digit DID numbers start with **501** to SIP domain **avayalab.com** (after being translated by the Avaya SBCE from the service provider public IP address **64.xxx.xxx.174**). The Dial Pattern uses the Route Policy **Inbound_To_car2_cores** as defined in **Section 6.77**. These DID numbers are assigned to the enterprise by Windstream.



The second example shows the Dial Pattern for outgoing calls that 11-digit dialed numbers begin with digit **1**. The Dial Pattern uses Routing Policy **Outbound_To_Windstream** as defined in **Section 6.7** to route outgoing calls to the Avaya SBCE.



6.9. Add Avaya Aura ® Session Manager

The creation of a Session Manager element provides the linkage between System Manager and Session Manager. This was most likely done as part of the initial Session Manager installation. To add Session Manager, navigate to **Home → Elements → Session Manager → Session Manager Administration** in the left navigation pane and click on the **New** button in the right pane (not shown). If Session Manager already exists, click **View** (not shown) to view the configuration. Enter or verify the data as described below and shown in the following screen:

In the **General** section, enter the following values:

- **SIP Entity Name**: Select the SIP Entity created for Session Manager.
- **Description**: Add a brief description (optional).
- **Management Access Point Host Name/IP**: Enter the IP address of the Session Manager management interface.

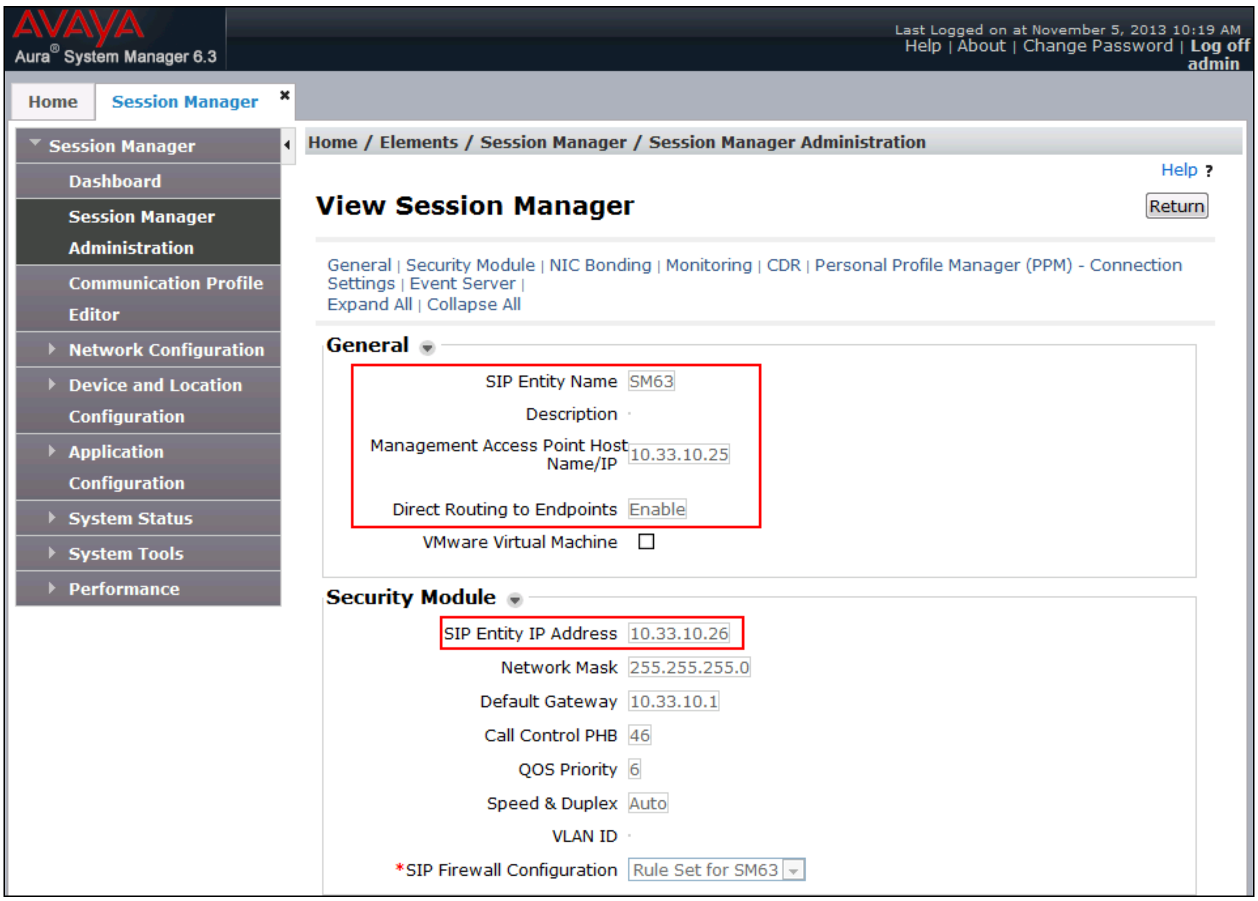
In the **Security Module** section, enter the following values:

- **SIP Entity IP Address**: Should be filled in automatically based on the SIP Entity Name. Otherwise, enter IP address of Session Manager signaling interface.
- **Network Mask**: Enter the network mask corresponding to the IP address of Session Manager.
- **Default Gateway**: Enter the IP address of the default gateway for Session Manager.

In **Monitoring** section, verify **Enable Monitoring** is checked.

Use default values for the remaining fields. Then click **Save** (not shown).

The screenshots below show Session Manager values.



7. Configure Avaya Session Border Controller for Enterprise

This section covers the configuration of the Avaya SBCE. It is assumed that the software has already been installed. For additional information on these configuration tasks, see **References** Error! Reference source not found. and Error! Reference source not found. in **Section 11**.

The compliance testing comprised the configuration for two major components, Trunk Server for service provider and Call Server for enterprise. Each component consists of a set of Global Profiles, Domain Policies and Device Specific Settings. The configuration is defined in the Avaya SBCE web user interface as described in the following sections.

Trunk Server configuration elements for service provider Windstream:

- Global Profiles:
  - URI Groups
  - Routing
  - Topology Hiding
  - Server Interworking
  - Signaling Manipulation
  - Server Configuration
- Domain Policies:
  - Application Rules
  - Media Rules
  - Signaling Rules
  - Endpoint Policy Group
  - Session Policy
- Device Specific Settings:
  - Network Management
  - Media Interface
  - Signaling Interface
  - End Point Flows → Server Flows
  - Session Flows

Call Server configuration elements for enterprise Session Manager:

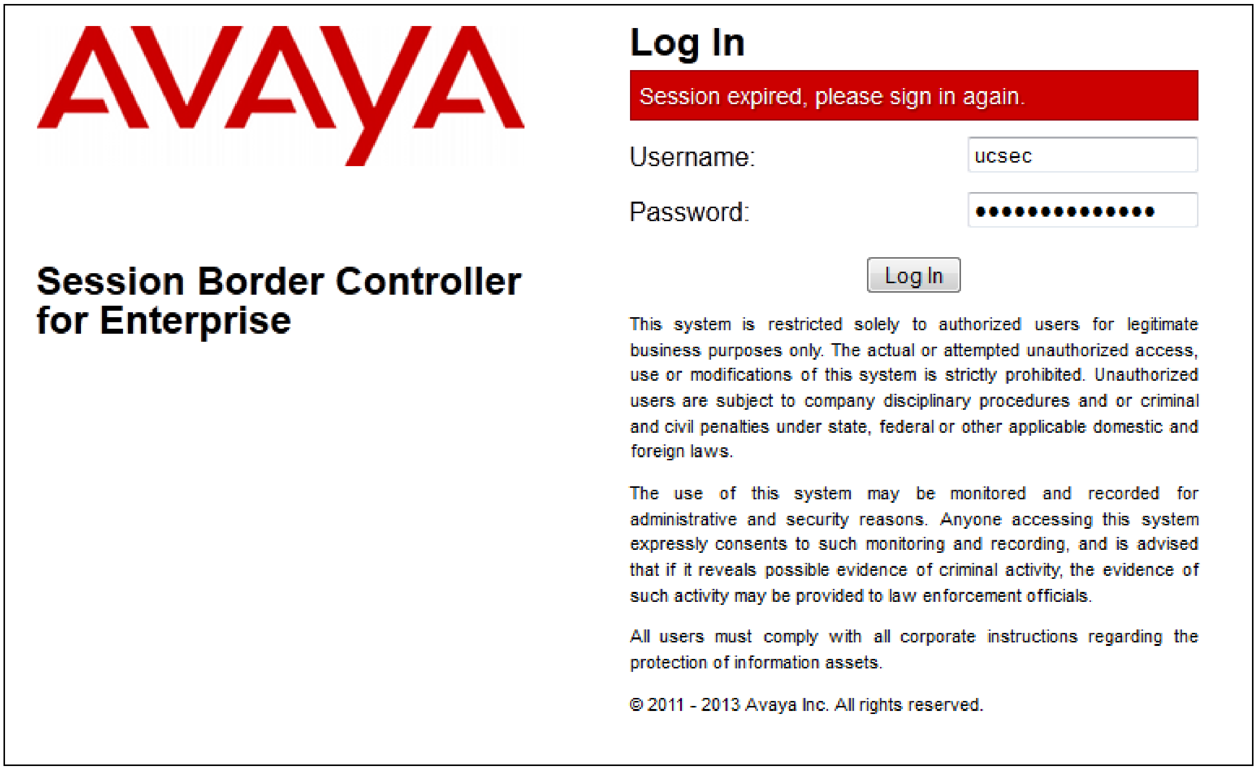
- Global Profiles:
  - URI Groups
  - Routing
  - Topology Hiding
  - Server Interworking
  - Server Configuration
- Domain Policies:
  - Application Rules
  - Media Rules
  - Signaling Rules
  - Endpoint Policy Group
  - Session Policy
- Device Specific Settings:

- Network Management
- Media Interface
- Signaling Interface
- End Point Flows → Server Flows
- Session Flows

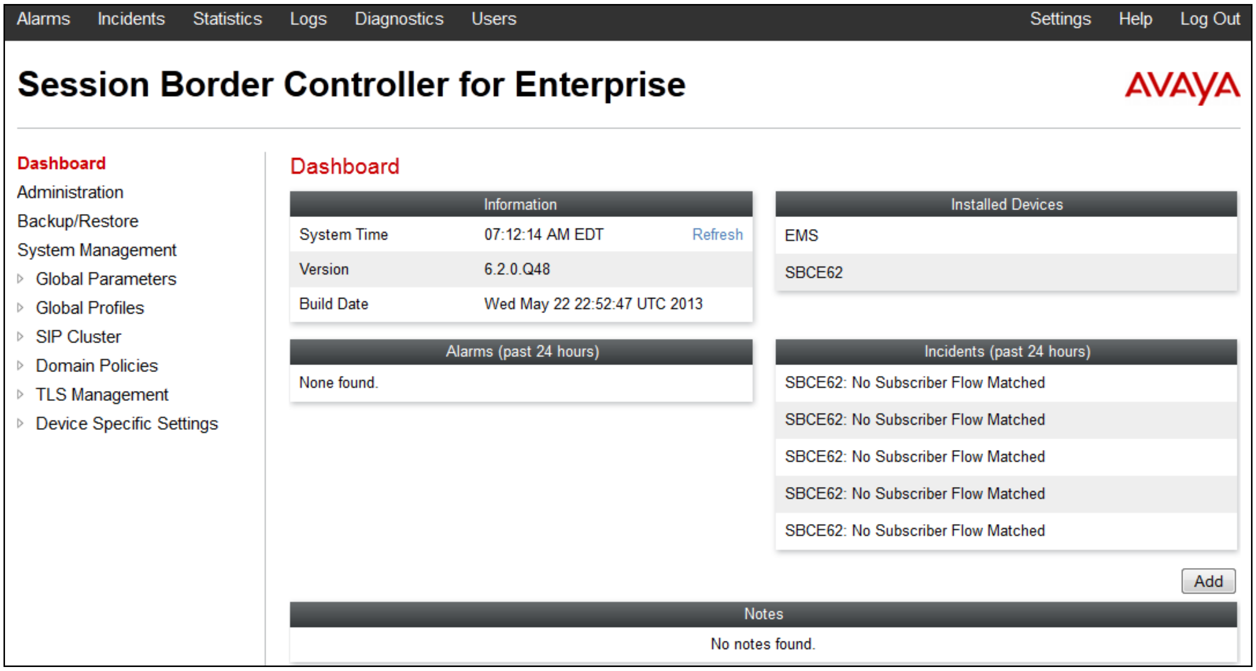
7.1. Log into Avaya Session Border Controller for Enterprise

Use a web browser to access Avaya Session Border Controller for Enterprise (Avaya SBCE) web interface, enter "**https://<ip-addr>/sbc**" in the address field of the web browser, where <ip-addr> is the management LAN IP address of ASBCE (not shown).

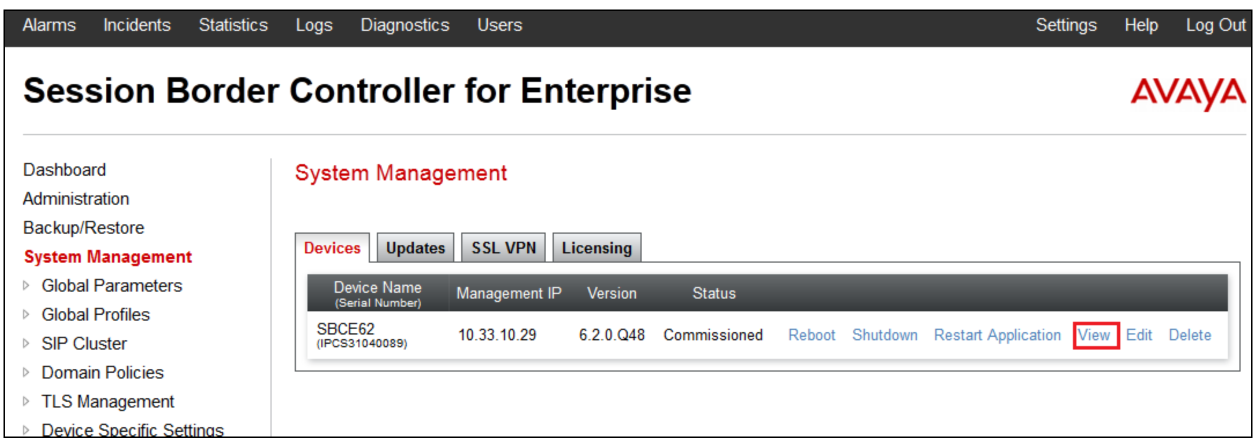
Enter the appropriate credentials then click **Log In**.



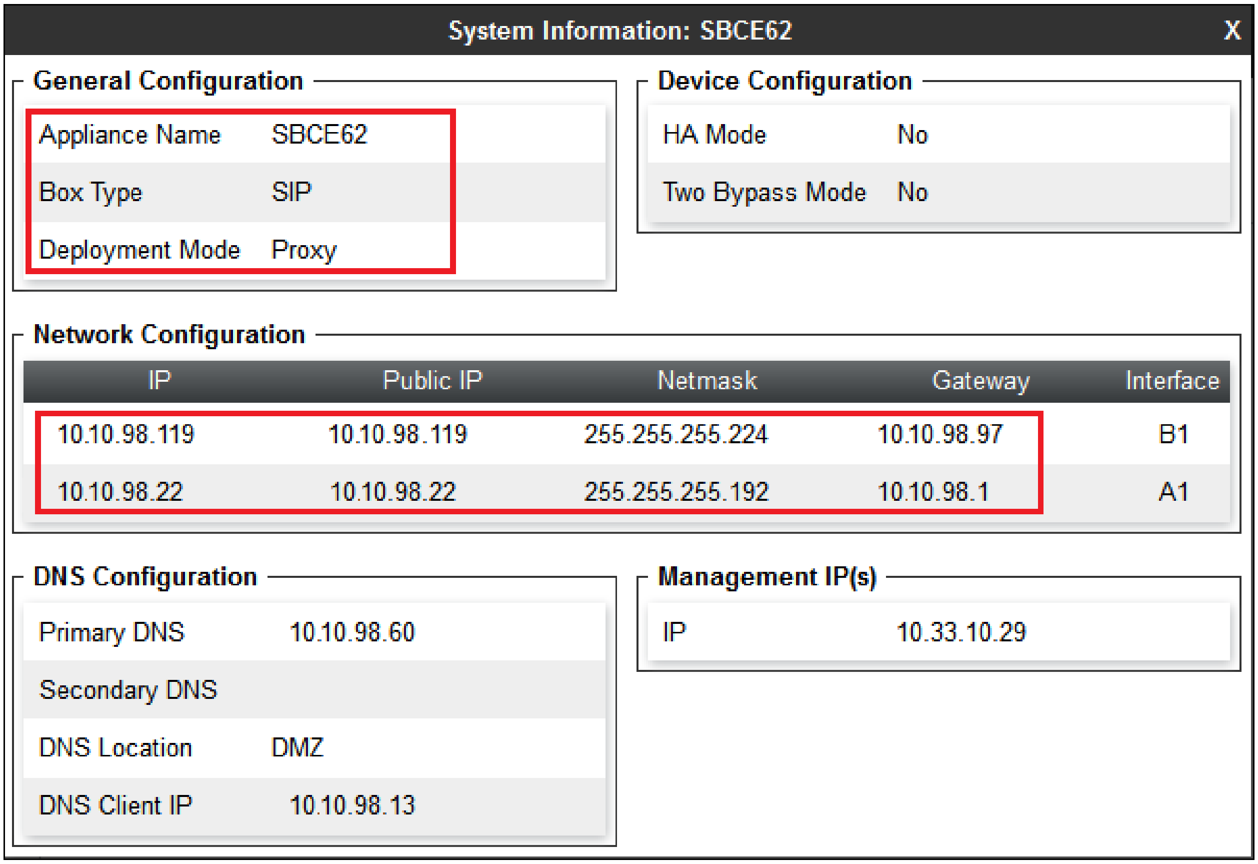
The main page of the Avaya SBCE will appear as shown below.



To view system information that has been configured during installation, navigate to **System Management** from the left menu pane. A list of installed devices is shown in the right pane. In the Compliance test, a single device named **SBCE62** is added. To view the configuration of this device, click the **View** link as shown below.



The **System Information** screen shows **General Configuration**, **Device Configuration**, **Network Configuration, DNS Configuration** and **Management IP** information provided during installation and corresponds to **Figure 1**. The **Box Type** is set to **SIP** and the **Deployment Mode** is set to **Proxy**. Default values are used for all other fields.



7.2. Global Profiles

Global Profiles allows for configuration of parameters across all Avaya SBCE.

7.2.1. Uniform Resource Identifier (URI) Groups

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. These groups are used by the various domain policies to determine which actions (Allow, Block, or Apply Policy) should be used for a given call flow.

To add an URI Group, select **System Management → Global Profiles → URI Groups** and click on the **Add** button.

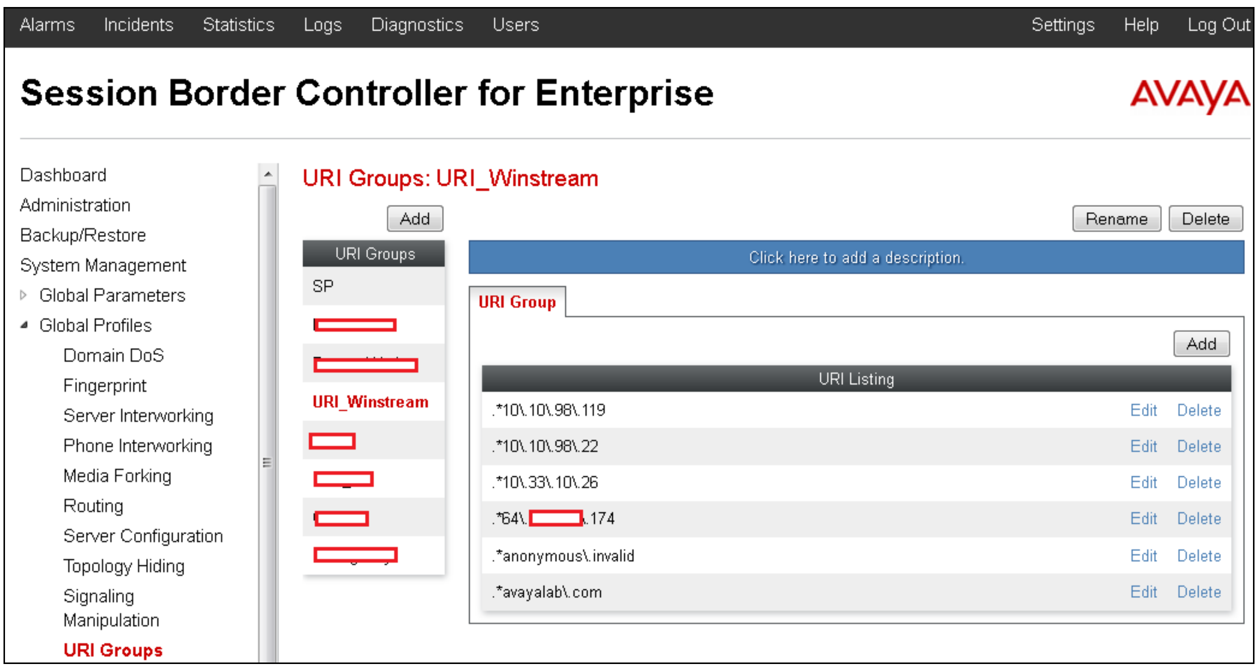
In the compliance testing, a URI Group named **URI_Windstream** was added with following URI type **Regular Expression**:

- "**.*10\.33\.10\.26**" – the IP addresses of URI-Host in OPTIONS heartbeat originated by Session Manager.

- "**.*10\.10\.98\.119**" – the public IP address of ASBCE.
- "**.*.10.\10.98\.22**" – the internal IP address of ASBCE.
- "**.*10\.1\.6.\.110**" – the private IP address of the service provider.
- "**.*64\.xxx\.xx\.174**" – the public proxy IP address of the service provider.
- "**.*anonymous\.invalid**" – the anonymous domain for the private call.
- "**.*avayalab\.com**" – the enterprise SIP domain.

This URI-Group is used to match the "From" and "To" headers in a SIP call dialog received from both the CS1000 and Windstream. If there is a match, the Avaya SBCE will apply the appropriate Routing profile (see **Section 7.2.2**) and Server Flow (see **Section 7.4.4**) to route incoming and outgoing calls to the right destination.

The screenshot below illustrates the URI listing for URI Group **URI_Windstream**.



7.2.2. Routing Profiles

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 determine 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.

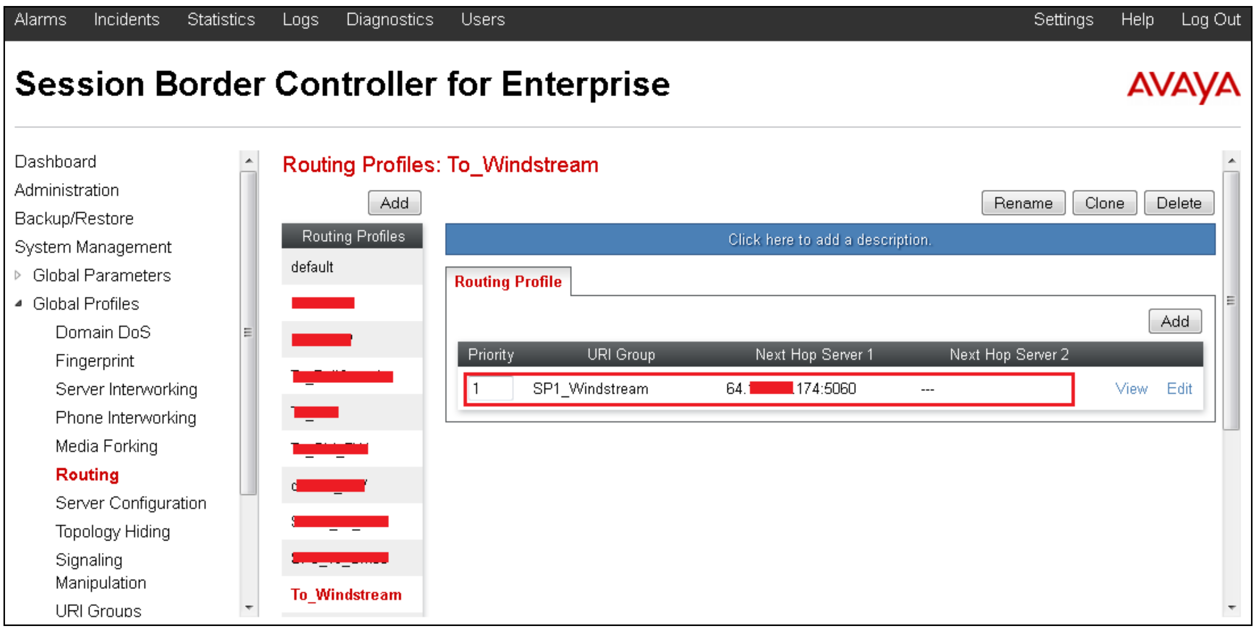
To create a Routing profile, select **System Management → Global Profiles → Routing** then click on the **Add** button.

In the compliance testing, a Routing profile **To_Windstream** was created to be used in conjunction with the Server Flow (see **Section 7.4.4**) defined for the CS1000. This entry is used to route outgoing calls from the enterprise to Windstream.

In the opposite direction, a Routing profile **To_SM63_CAR276** was created to be used in conjunction with the Server Flow (see **Section 7.4.4**) defined for Windstream. This entry is used to route incoming calls from Windstream to the enterprise.

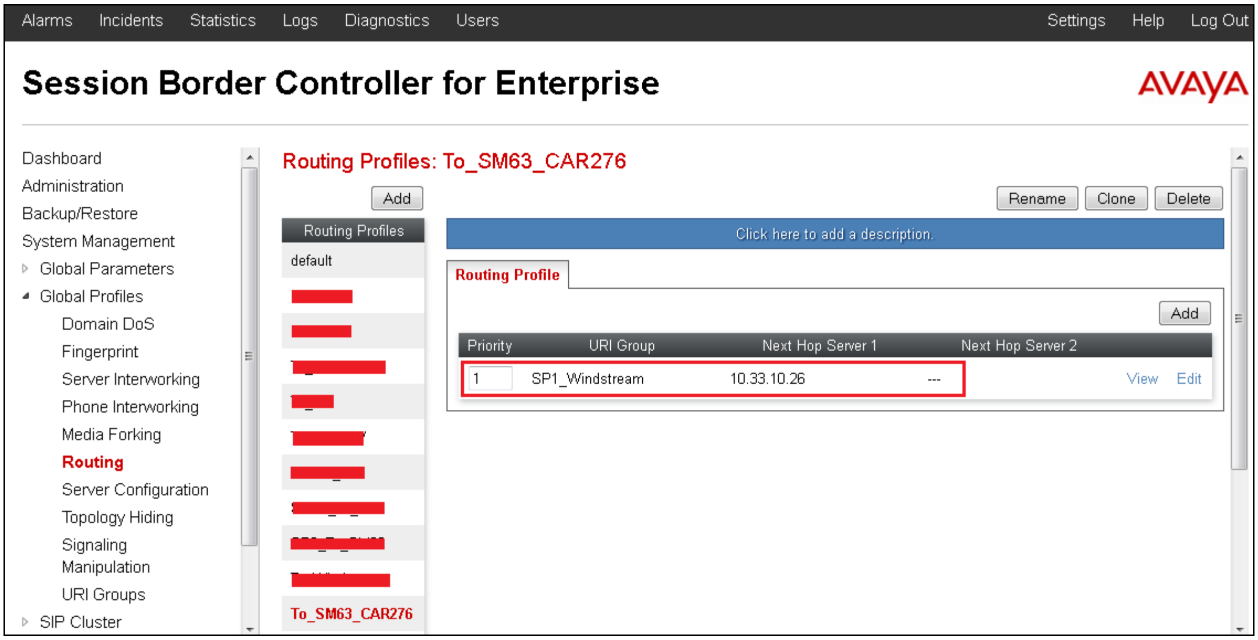
7.2.2.1 Routing Profile for Windstream

The screenshot below illustrates the **System Management → Global Profiles → Routing**: **To_Windstream**. If there is a match between the SIP domain in the "To" header with the URI Group **SP1_Windstream** defined in **Section 7.2.1**, the call will be routed to the **Next Hop Server 1** which is the proxy IP address of Windstream Trunk Server on port **5060**.



7.2.2.2 Routing Profile for Avaya Aura ® Session Manager

The Routing Profile **To_SM63_CAR276** in the screenshot below was defined to route calls where the SIP domain in the "To" header matches the URI-Group **SP1_Windstream** defined in **Section 7.2.1**, to **Next Hop Server 1** which is the IP address of Session Manager on port 5060.



7.2.3. Topology Hiding

Topology Hiding is a security feature of the Avaya SBCE which allows changing certain key SIP message parameters to 'hide' or 'mask' how the enterprise network may appear to an unauthorized or malicious user.

To create a Topology Hiding profile, select **System Management → Global Profiles → Topology Hiding** then click on **Add**.

In the compliance testing, two Topology Hiding profiles were created: **Topo_Windstream** and **Topo_4_CAR276**.

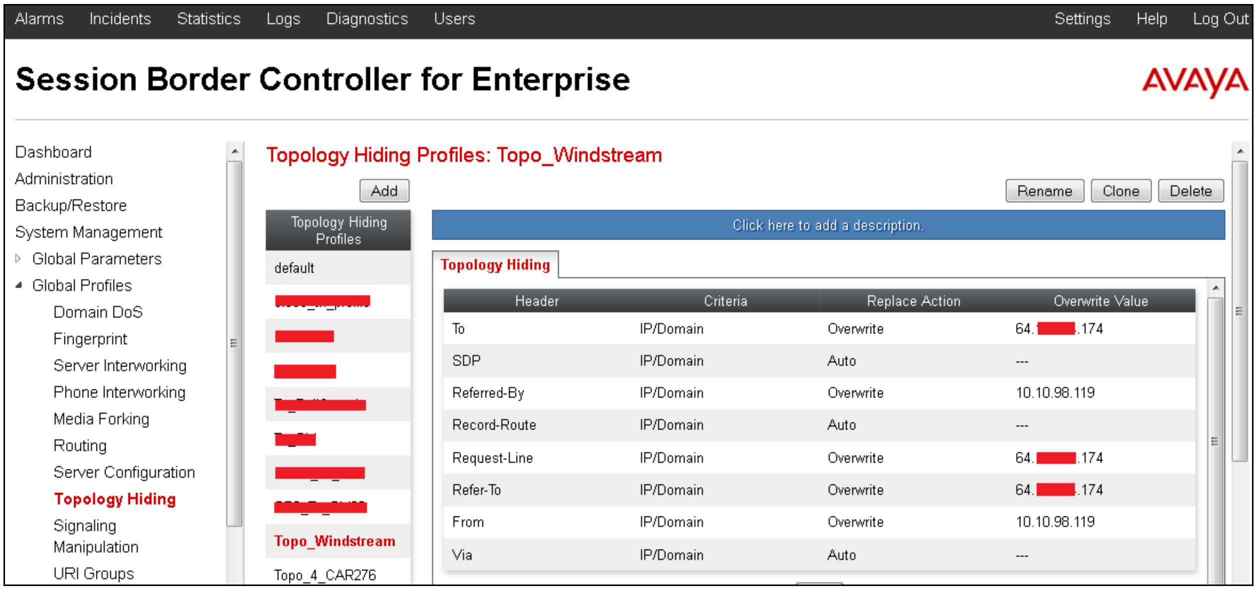
7.2.3.1 Topology Hiding Profile for Windstream

Topology Hiding profile **Topo_Windstream** was defined for outgoing calls to Windstream to:

- Mask URI-Host of the "Request-Line" and "To" headers with service provider public IP address **64.xxx.xx.174** to meet the requirements of Windstream.
- Change the "From" header added by the CS1000 with external IP address of ASBCE known to Windstream.

This implementation is to secure the enterprise network topology and also to meet the SIP requirements from the service provider.

The screenshots below illustrate the Topology Hiding profile **Topo_Windstream**

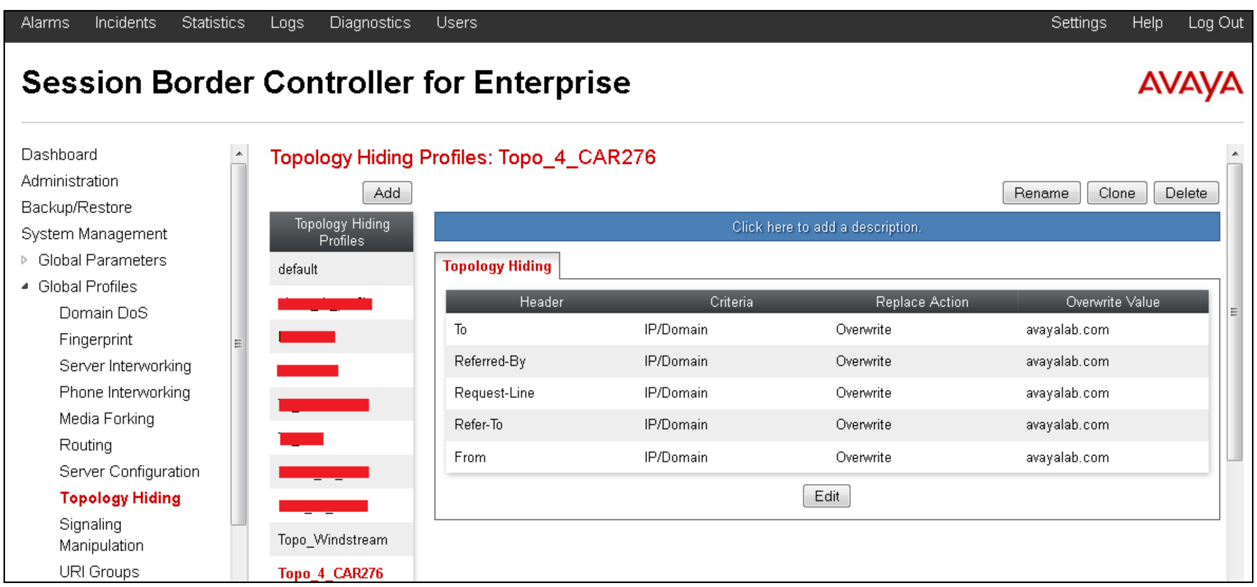


7.2.3.2 Topology Hiding Profile for the CS1000

Topology Hiding profile **Topo_4_CAR276** was defined for incoming calls to the CS1000 to:

- Mask URI-Host of the "Request-Line", "To", "Refer-By", "Refer-To" and "From" headers with the enterprise SIP domain **avayalab.com**.

The screenshots below illustrate the Topology Hiding profile **Topo_4_CAR276**.



Note:

- The **Criteria** should be **IP/Domain** to allow the ASBCE to mask both domain name and IP address presenting in the URI-Host.

7.2.4. Server Interworking

Server Interworking profile features are configured differently for Call Server and Trunk Server. To create a Server Interworking profile, select **System Management → Global Profiles → Server Interworking** then click on the **Add** button (not shown).

In the compliance testing, two Server Interworking profiles **Inter_Windstream** and **Inter_CAR276** were created for Windstream (Trunk Server) and the CS1000 (Call Server).

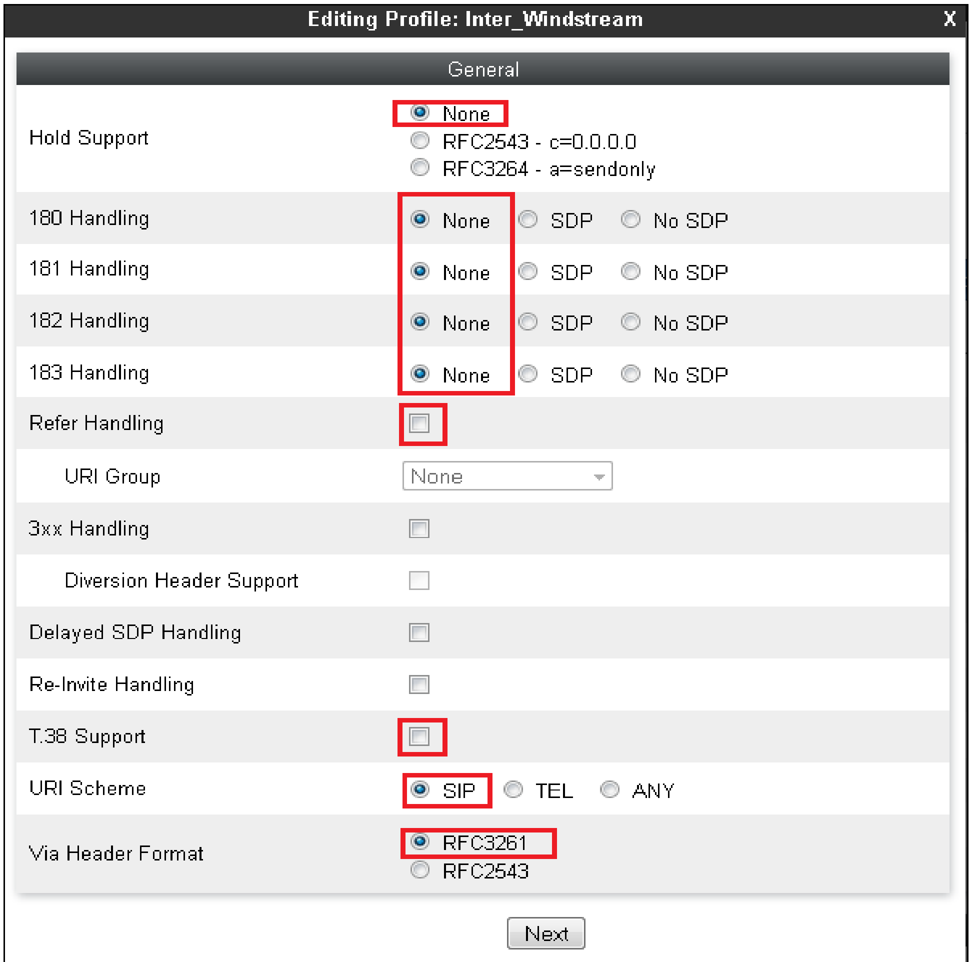
7.2.4.1 Server Interworking Profile for Windstream

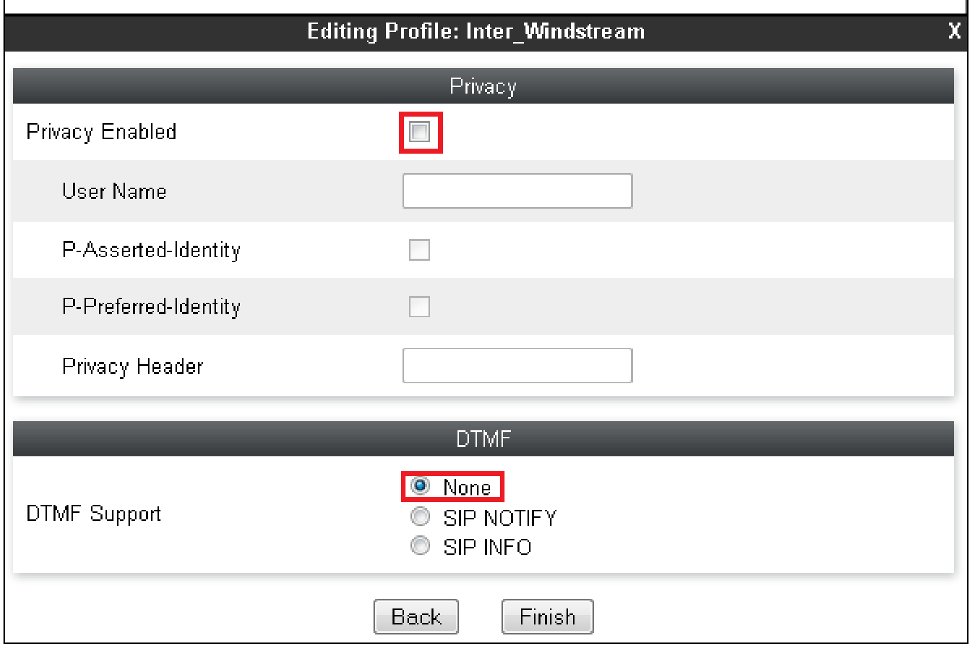
Server Interworking profile **Inter_Windstream** was defined to match the specification of Windstream. The **General** and **Advanced** tabs were configured with the following parameters while the other tabs **Timers, URI Manipulation** and **Header Manipulation** were kept as default.

General settings:

- **Hold Support** = **None**. The Avaya SBCE will not handle Hold/ Resume signaling, it keeps the Hold/ Resume signaling unchanged to send to the destination server.
- **18X Handling** = **None**. The ASBCE will not handle 18X, it keeps the incoming 18X responds unchanged to send to the destination server.
- **Refer Handling** = **Unchecked**. The ASBCE will not handle Refer, it keeps REFER unchanged to send to the destination server.
- **T.38 Support** = **Unchecked**. Windstream does not support the T.38 codec for fax over IP in the compliance testing.
- **URI Scheme = select SIP**.
- **Via Header Format = select RFC3261**.
- **Privacy Enabled** = **Unchecked**. The ASBCE will not mask the "From" header with **anonymous** to the destination server. It depends on the far end to enable/ disable the "Privacy" on individual call basis.
- **DTMF Support** = **None**. The ASBCE will not modify the original DTMF transmission method sent by CS1000. It keeps the DTMF unchanged to send to the destination server.

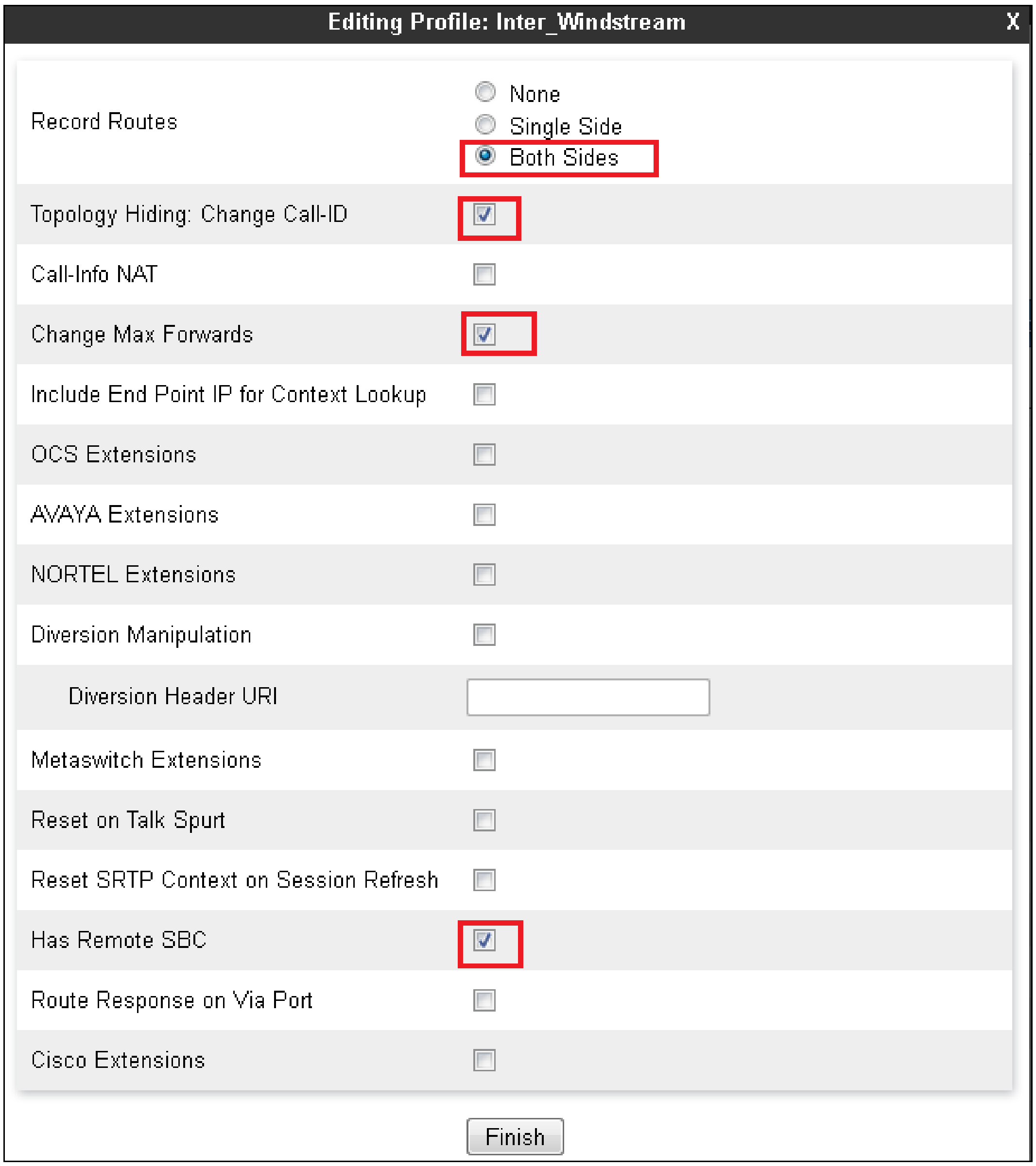
The screenshots below illustrate the Server Interworking profile **Inter_Windstream**.





Advanced settings:

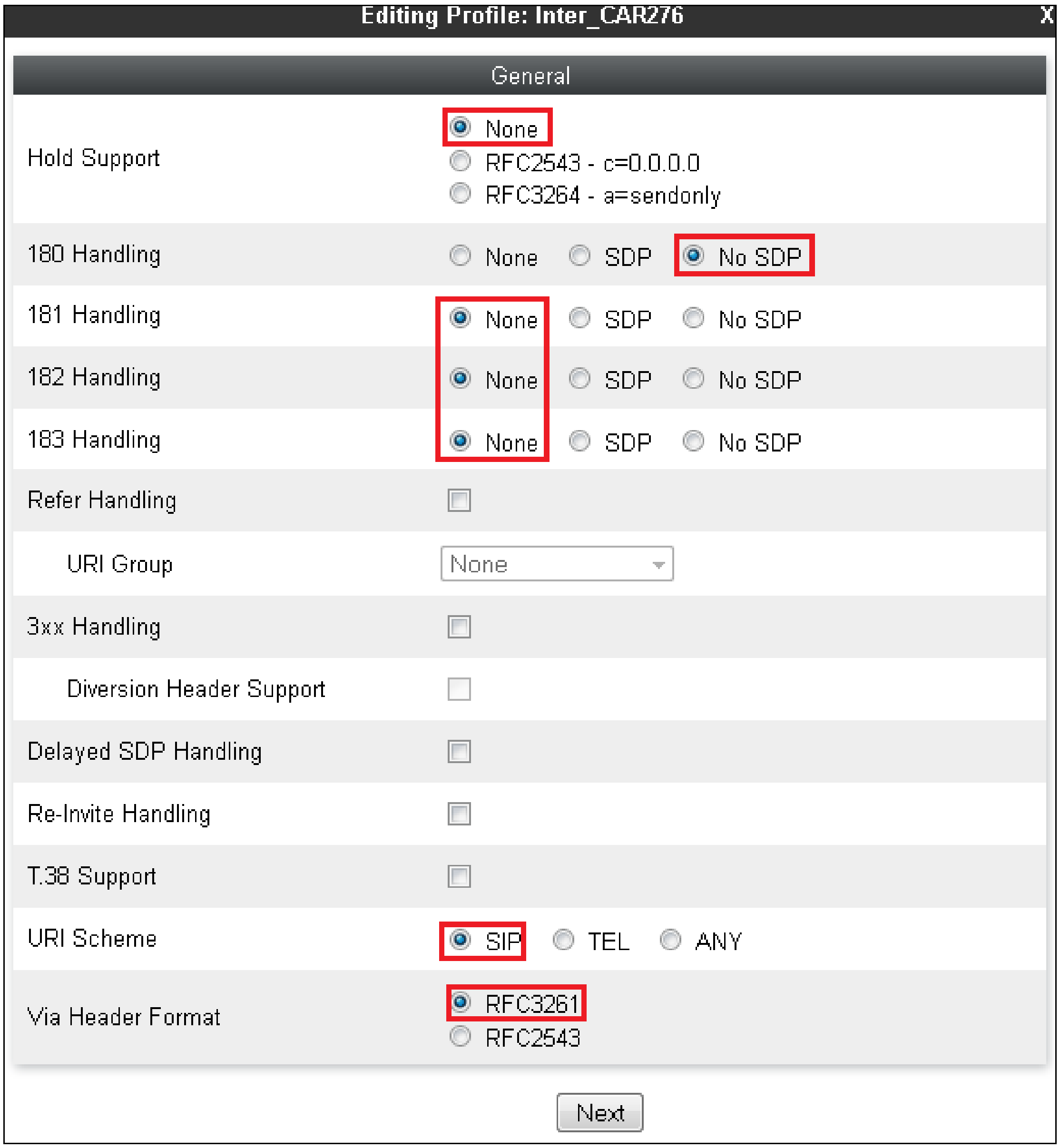
- **Record Routes** = **Both Sides**. The ASBCE will send the "Record-Route" header to both the CS1000 and Windstream.
- **TopologyHiding**: **Change Call-ID** = **Checked**. The ASBCE will mask the "Call-ID" header for the calls to the destination server.
- **Change MaxForwards** = **Checked**. The ASBCE will reduce the counter of the "Max-Forwards" header by 1 for the calls to the destination server.
- **Has Remote SBC** = **Checked**. The ASBCE will flexibly handle the changes to the SDP when the call is active.

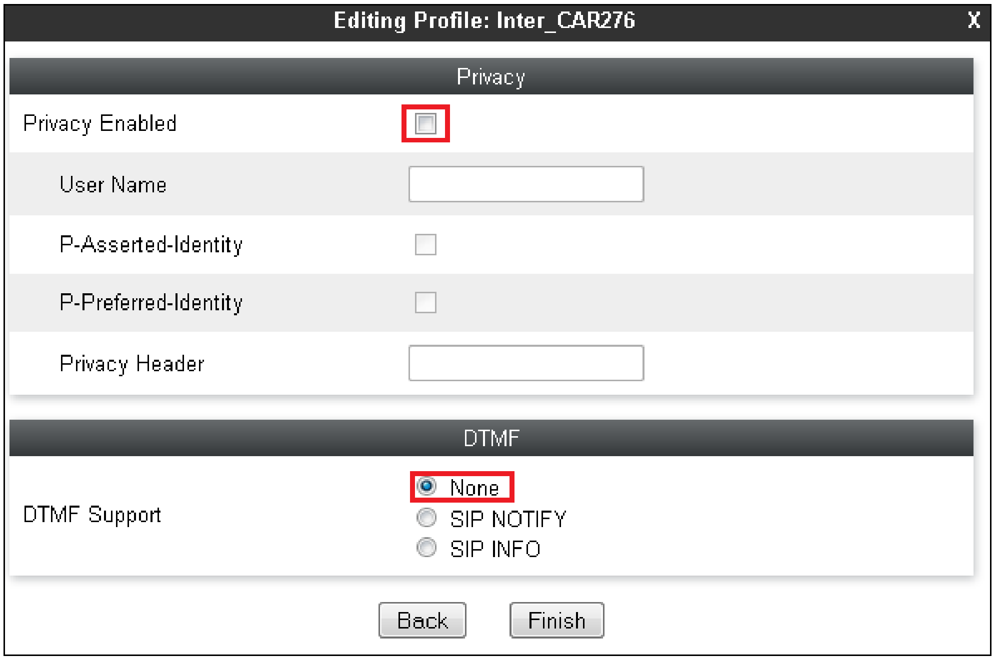


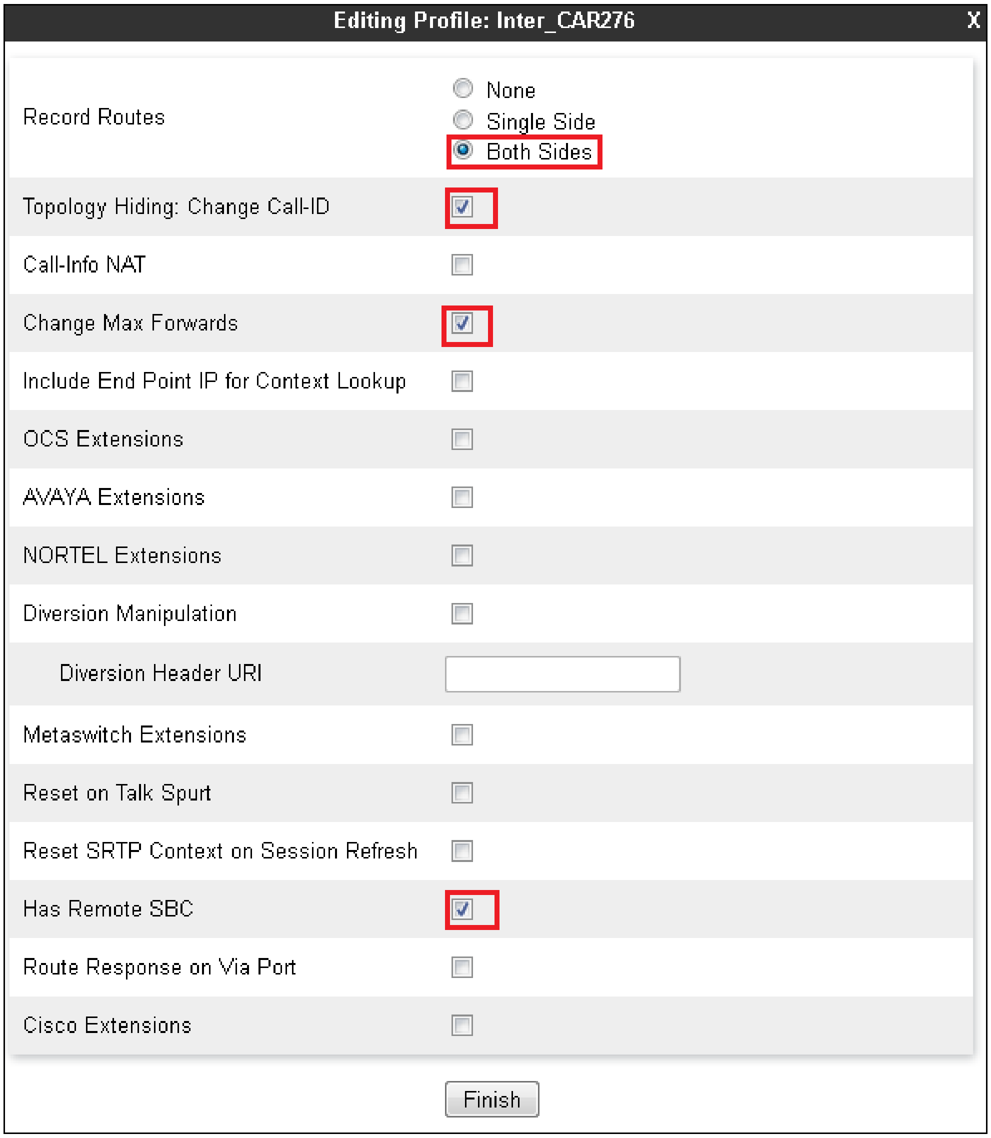
7.2.4.2 Server Interworking Profile for the CS1000

Server Interworking profile **Inter_CAR276** was similarly defined to match the specification of the CS1000. The "**180 Handling**" field must set to "**No SDP**" to fix the off-net blind transfer issue.

The screenshots below illustrate the Server Interworking profile **Inter_CAR276**.




Editing Profile: Inter_CAR276
Privacy
Privacy Enabled
User Name
P-Asserted-Identity
P-Preferred-Identity
Privacy Header
DTMF
DTMF Support
None
SIP NOTIFY
SIP INFO
Back
Finish



7.2.5. Signaling Manipulation

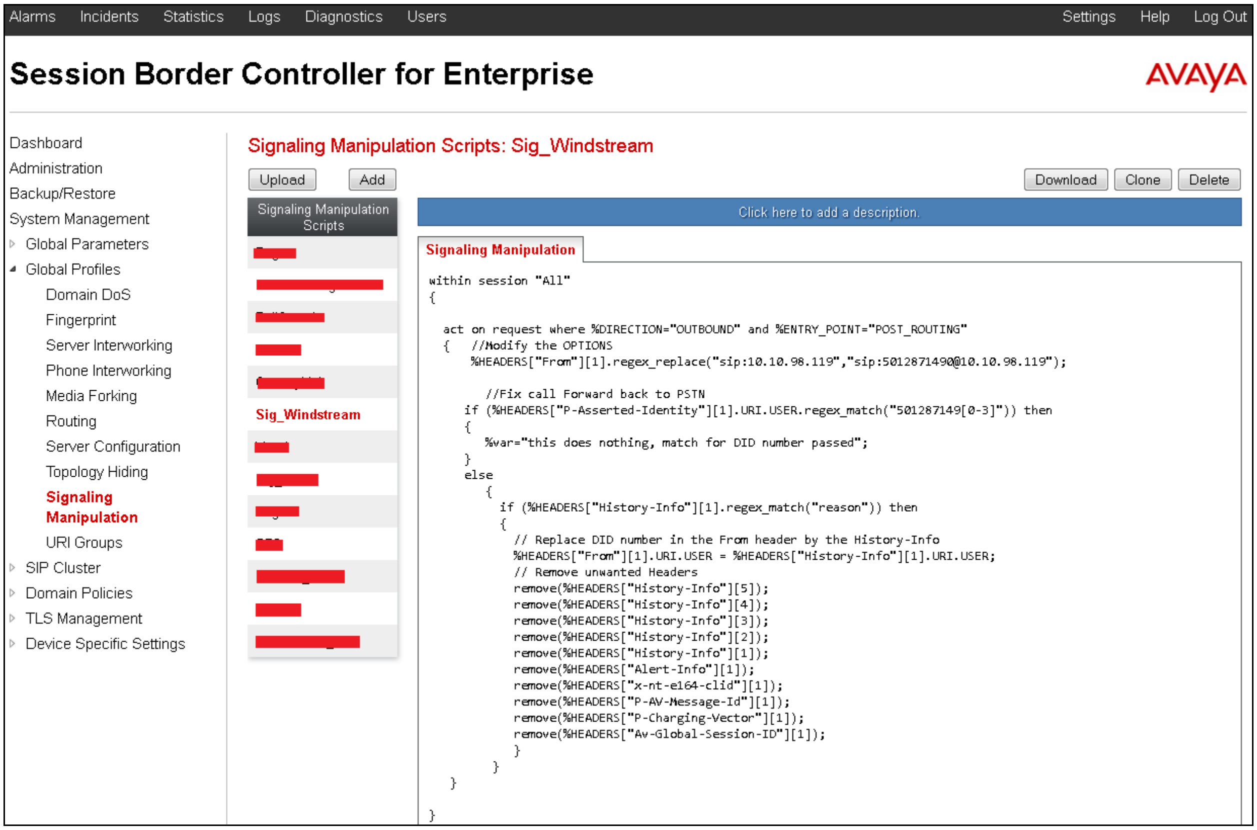
Signaling Manipulation feature allows the ability to add, change and delete any of the headers in a SIP message. This feature will add the ability to configure such manipulation in a highly flexible manner using a proprietary scripting language called SigMa.

The SigMa scripting language is designed to express any of the SIP header manipulations done by the ASBCE. Using this language, a script can be written and tied to a given Server Configuration (see **Section 0**) through the SBC web interface. The ASBCE then interprets this script at the given entry point or "hook point".

These Application Notes will not discuss the full feature of the Signaling Manipulation but will show an example of a script created during compliance testing to aid in Topology Hiding.

To create a Signaling Manipulation script, select **System Management → Global Profiles → Signaling Manipulation** then click on the **Add** button.

In the compliance testing, a SigMa script named **Sig_Windstream** was created for Server Configuration for Windstream and described detail as following:



The statement **"act on request where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING""** is to specify the script will take effect on all types of SIP messages for outbound calls to Windstream and the manipulation will be done on the header of the OPTIONS message to change from sip:10.10.98.119 to sip:501xxx1490@10.10.98.119 which is DID 501xxx1490 known by Windstream.

```
act on request where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
{
 //Modify the OPTIONS
%HEADERS["From"][1].regex_replace("sip:10.10.98.119","sip:501xxx1490@10.10.98.119");
```

The script below is to fix the call forward from PSTN back to PSTN and to remove unwanted headers in the call forward.

```
//Fix call Forward back to PSTN
     if (%HEADERS["P-Asserted-Identity"][1].URI.USER.regex_match("501287149[0-3]"))
then
     {
        %var="this does nothing, match for DID number passed";
     }
     else
        {
          if (%HEADERS["History-Info"][1].regex_match("reason")) then
          {
            // Replace DID number in the From header by the History-Info
            %HEADERS["From"][1].URI.USER = %HEADERS["History-Info"][1].URI.USER;
            // Remove unwanted Headers
            remove(%HEADERS["History-Info"][5]);
            remove(%HEADERS["History-Info"][4]);
            remove(%HEADERS["History-Info"][3]);
            remove(%HEADERS["History-Info"][2]);
            remove(%HEADERS["History-Info"][1]);
            remove(%HEADERS["Alert-Info"][1]);
            remove(%HEADERS["x-nt-e164-clid"][1]);
            remove(%HEADERS["P-AV-Message-Id"][1]);
            remove(%HEADERS["P-Charging-Vector"][1]);
            remove(%HEADERS["Av-Global-Session-ID"][1]);
            }
```

7.2.6. Server Configuration

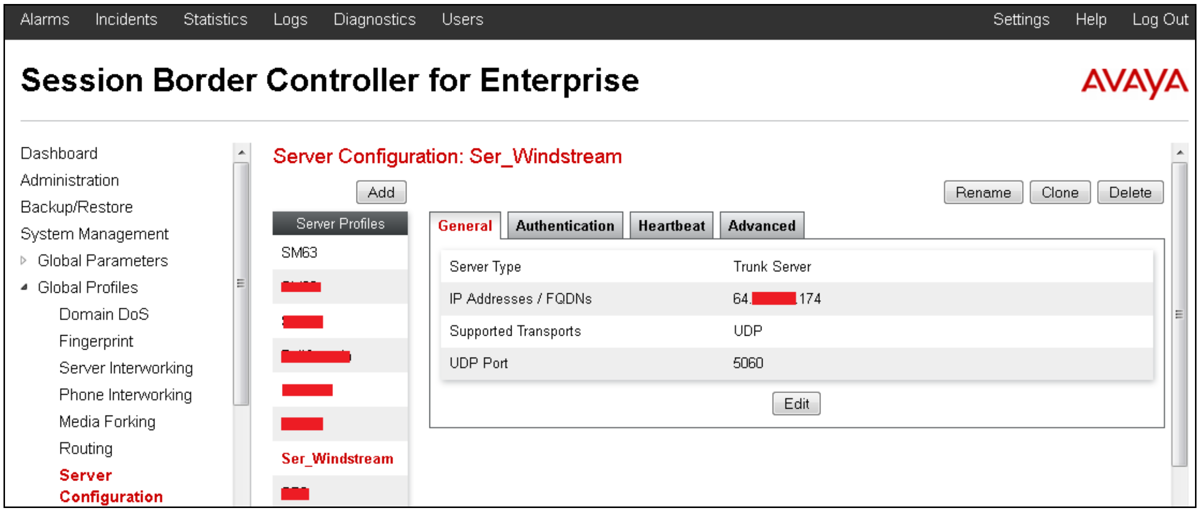
Server Configuration screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. These tabs are used to configure and manage various SIP Call Server specific parameters such as TCP and UDP port assignments, heartbeat signaling parameters, DoS security statistics and trusted domains.

To create a Server Configuration entry, select **System Management → Global Profiles →Server Configuration** then click on the **Add** button.

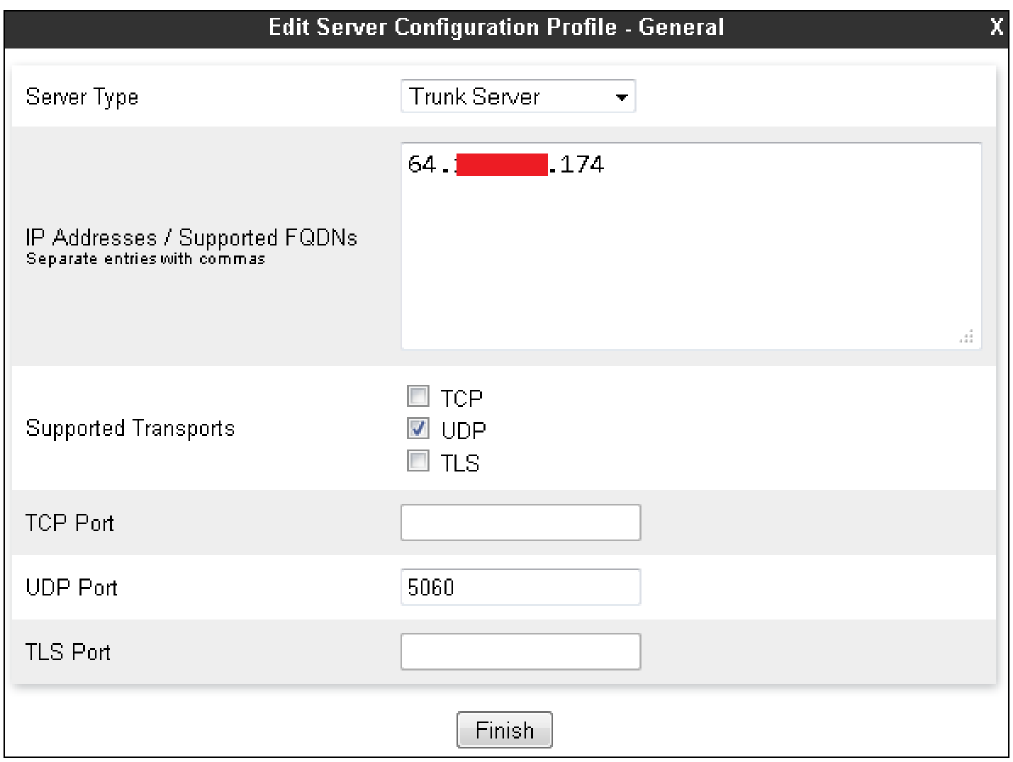
In the compliance testing, two separate Server Configurations were created, server entry **Ser_Windstream** for Windstream and server entry **SM63** for Session Manager.

7.2.6.1 Server Configuration for Windstream

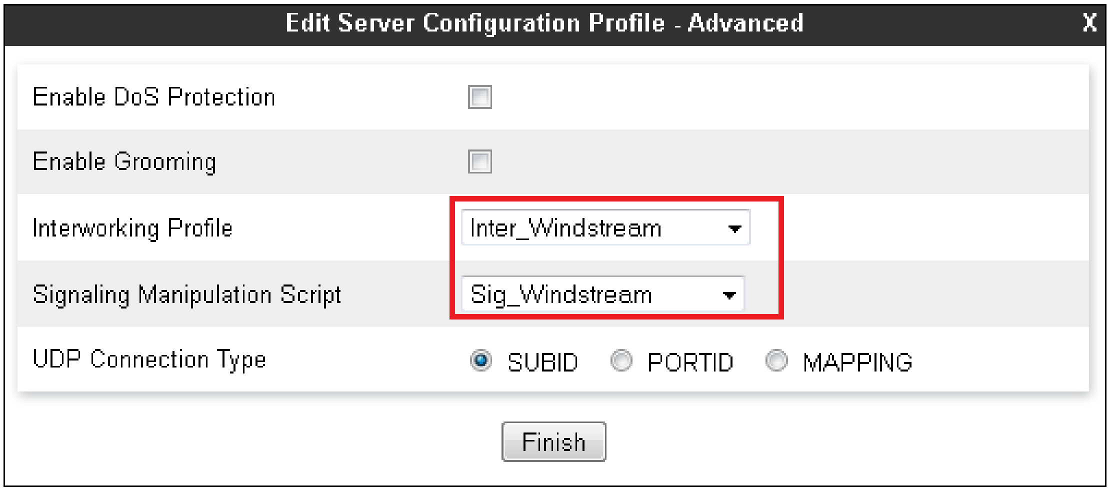
The Server Configuration **Ser_Windstream** was added for Windstream and is discussed in detail as shown below. The **General** and **Advanced** tabs were provisioned. The **Heartbeat** tab is kept as disabled as default to allow Avaya SBCE to forward OPTIONS message from Session Manager to Windstream. The screen below shows the server configuration for Windstream.



In the **General** tab, specify **Server Type** for Windstream as a **Trunk Server**. The IP connectivity has also been defined as shown in the screenshot below. In this compliance testing, Windstream supported transport protocol **UDP** and listens on port **5060**.

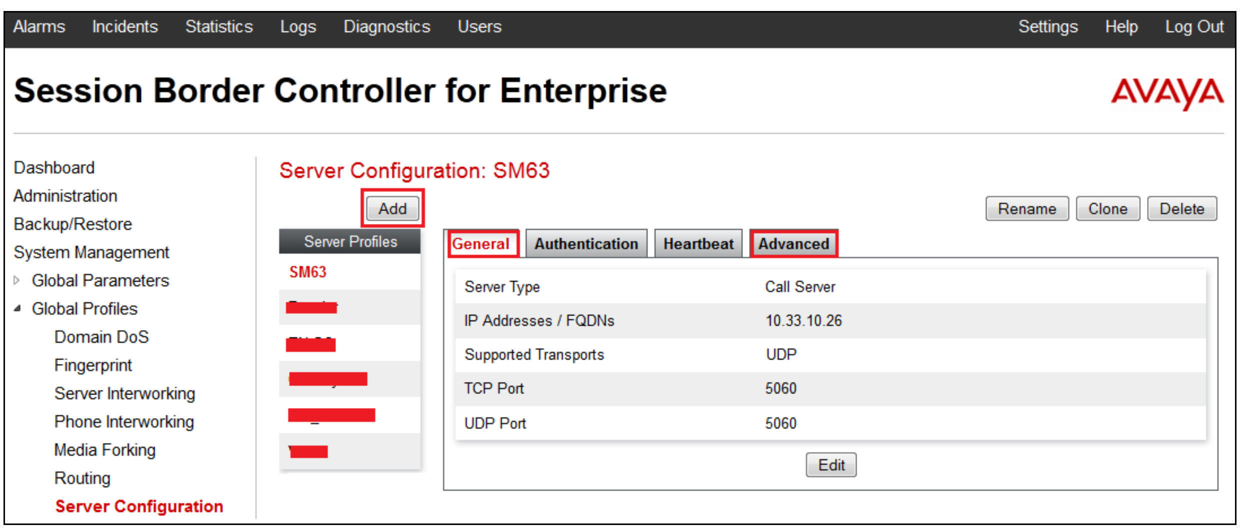


Under **Advanced** tab, for Interworking Profile drop down list, select **Inter_Windstream** as defined in **Section 7.2.4.1** and for **Signaling Manipulation Script** drop down list, select **Sig_Windstream** as defined in **Section 7.2.5**. These configurations are applied to the specific SIP profile and SigMa rules for the traffic from and to Windstream. The other settings are kept as default. Click **Finish** button to save and close the window.

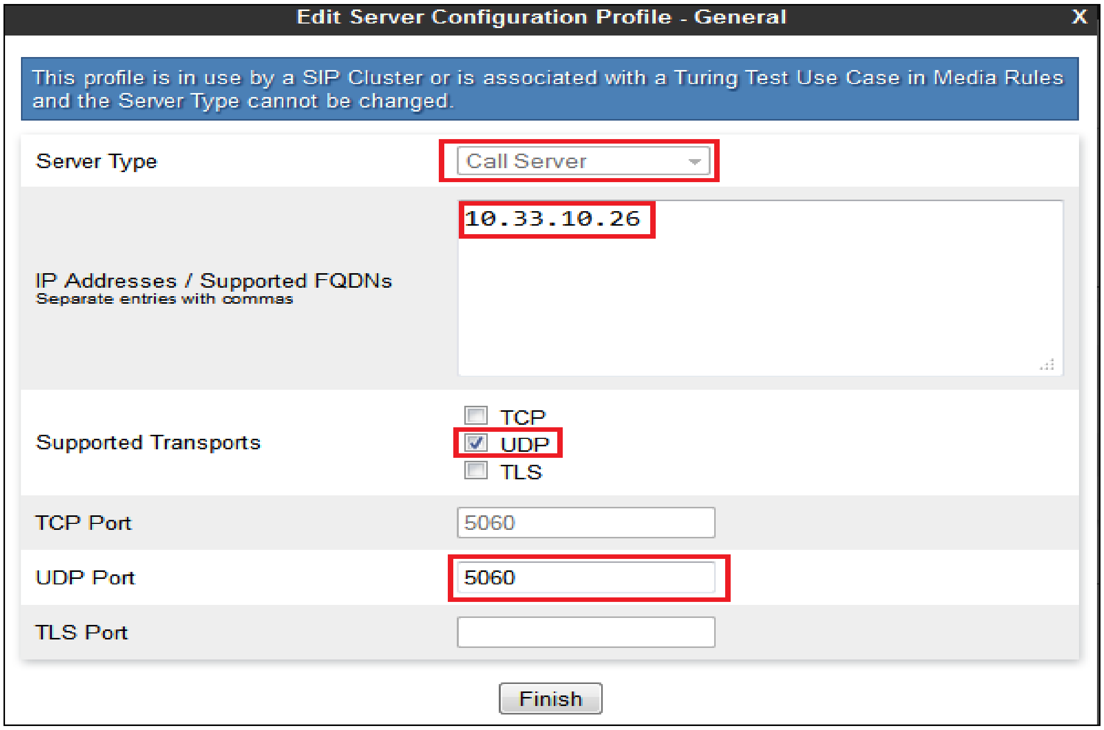


7.2.6.2 Server Configuration for Avaya Aura® Session Manager

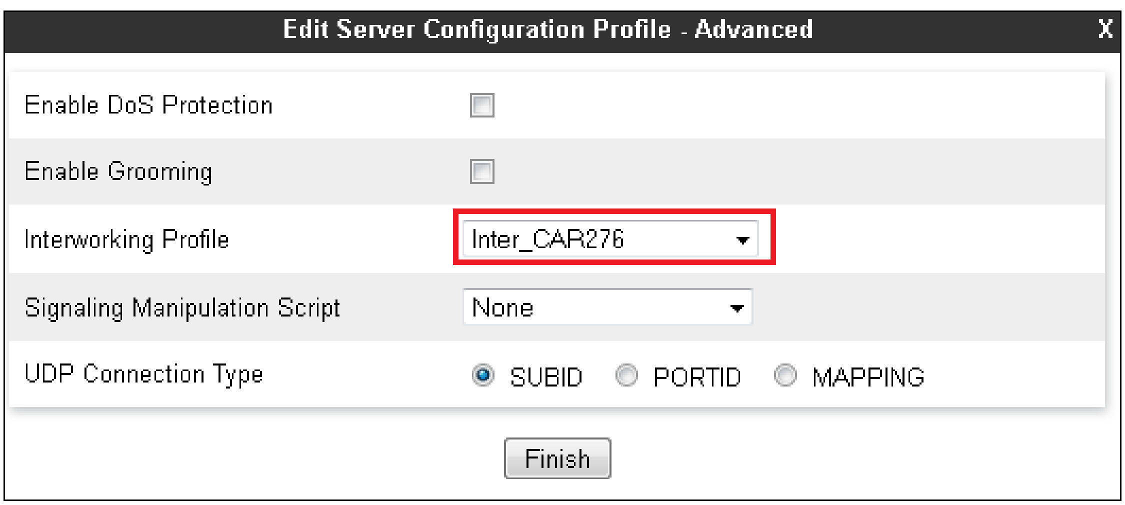
The Server Configuration **SM63** was added for Session Manager, it is discussed in detail as shown below. Only the **General** and **Advanced** tabs required provisioning. The **Heartbeat** tab is kept as disabled as default to allow the Avaya SBCE to forward the OPTIONS heartbeat from Windstream to Session Manager to query for the status of the SIP Trunk.



In the **General** tab, specify **Server Type** as **Call Server**. The IP connectivity has also been defined as shown in the screenshot below. In this compliance testing, Session Manager was configured with transport protocol **UDP** and listens on port **5060**. For detailed configuration, refer to **Section 6.6**. Click **Finish** button to save and close the window.



Under **Advanced** tab, for **Interworking Profile** drop down list, select **Intel_CAR276** as defined in **Section 7.2.4.2** and for **Signaling Manipulation Script** drop down list select **None**. The other settings are kept as default. Click **Finish** button to save and close the window.



7.3. Domain Policies

Domain Policies feature configures various rule sets (policies) to control unified communications based upon criteria of communication sessions originating from or terminating at the enterprise. These criteria can be used to trigger policies which, in turn, activate various security features of the ASBCE to aggregate, monitor, control and normalize call flow. There are default policies available for use, or a custom domain policy can be created.

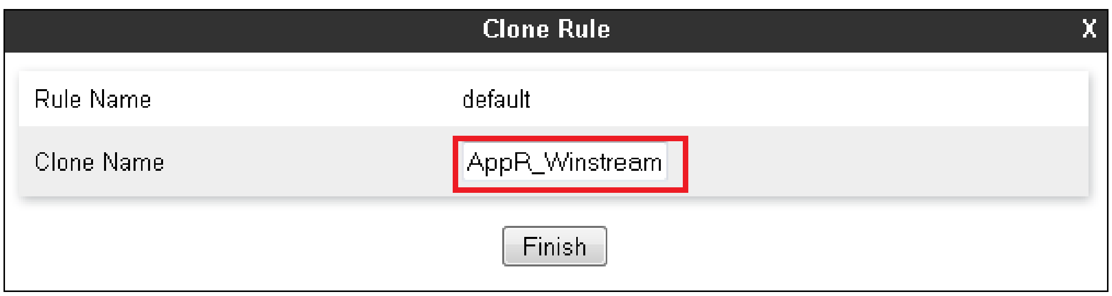
7.3.1. Application Rules

Application Rules define which types of SIP-based Unified Communications (UC) applications the ASBCE security device will protect: voice, video, and/or Instant Messaging (IM). In addition, it is possible to configure the maximum number of concurrent voice and video sessions the network will process in order to prevent resource exhaustion.

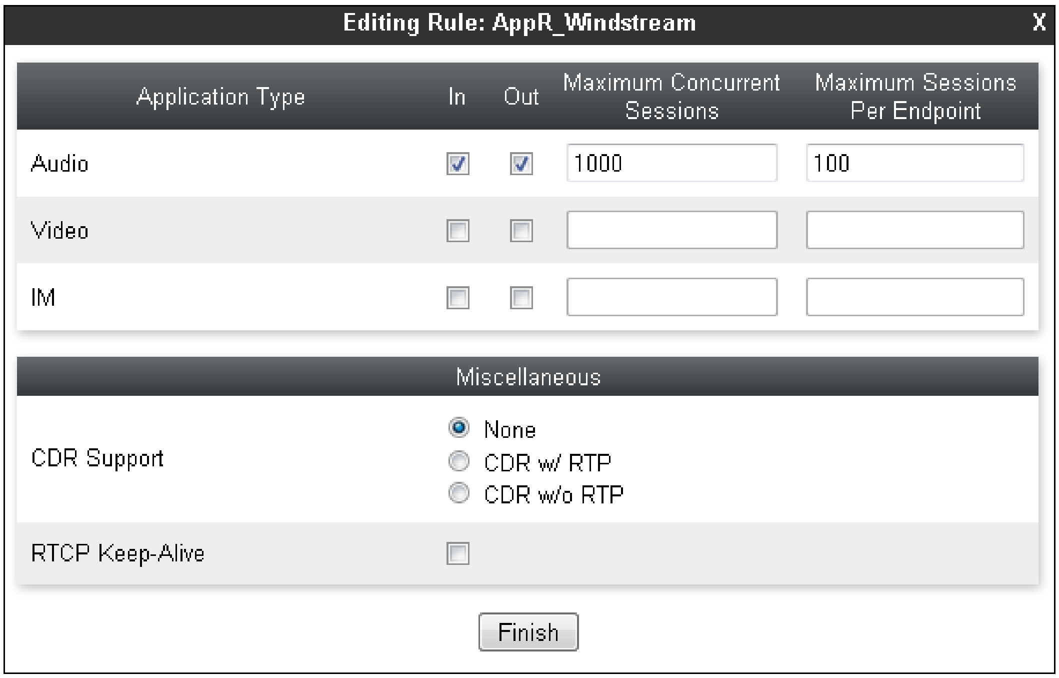
An Application Rule was created to set the number of concurrent voice traffic sessions. The sample configuration cloned and modified the default application rule to increase the number of **Maximum Concurrent Session** and **Maximum Sessions Per Endpoint**.

To clone an application rule, navigate to **Domain Policies → Application Rules**, select the default rule then click on the **Clone Rule** button (not shown).

Enter a descriptive name e.g. **AppR_Windstream** for the new rule then click on the **Finish** button.



Click **Edit** button (not shown) to modify the rule. Set the **Maximum Concurrent Sessions** and **Maximum Session Per Endpoint** for the **Audio** Application Type to a value high enough for the amount of traffic the network is able process. The following screen shows the modified Application Rule with the **Maximum Concurrent Sessions** and **Maximum Session Per Endpoint** set to **1000** and **100**. In the compliance testing, the CS1000 was programmed to control the concurrent sessions by setting the number of Virtual Trunks (see **Section 5.5.7**) to the allotted number. Therefore, the values in the Application Rule **AppR_Windstream** are set high enough to be considered non-blocking.



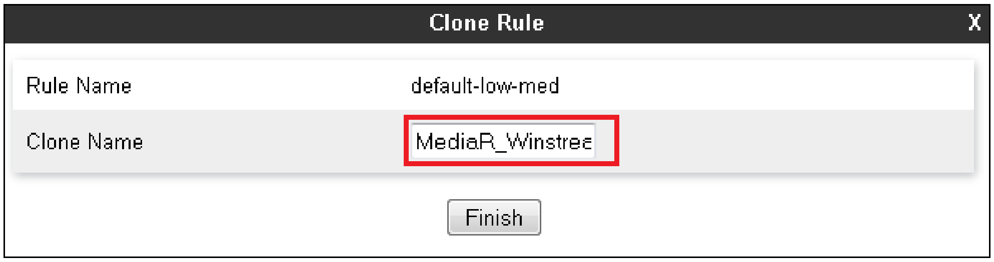
7.3.2. Media Rules

Media Rules 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 packet matching the criteria will be handled by the SBC security product.

A custom Media Rule was created to set the **Quality of Service** and **Media Anomaly Detection**. The sample configuration showed Media Rule **MediaR_Windstream** which was used for both the enterprise and Windstream networks.

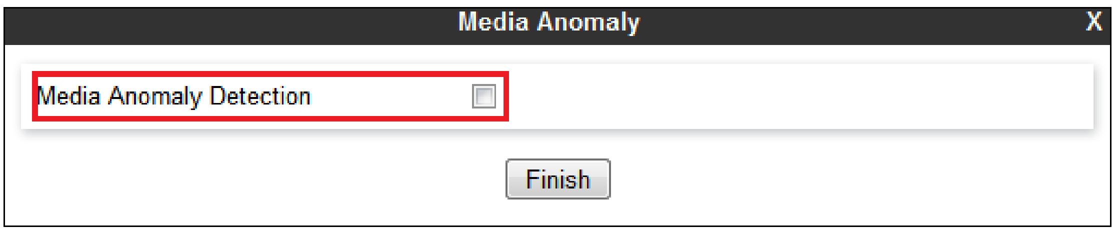
To create a **Media Rule**, navigate to **Domain Policies → Media Rules**, select the **default-low-med** rule then click on the **Clone** button (not shown).

Enter a descriptive name e.g. **MediaR_Windstream** for the new rule and then click **Finish** button.



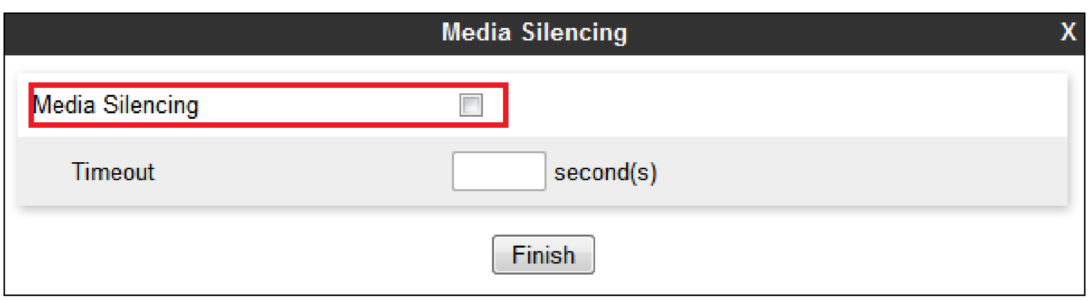
In the event of the RTP change, the ASBCE interprets this as an anomaly and an alert will be created in the **Incidents Log**. Disabling **Media Anomaly Detection** could prevent the **RTP Injection Attack** alerts from being created in the log when the audio attributes change.

To modify Media Anomaly, select the **Media Anomaly** tab and click on the **Edit** button (not shown). Then uncheck **Media Anomaly Detection** and click on the **Finish** button.

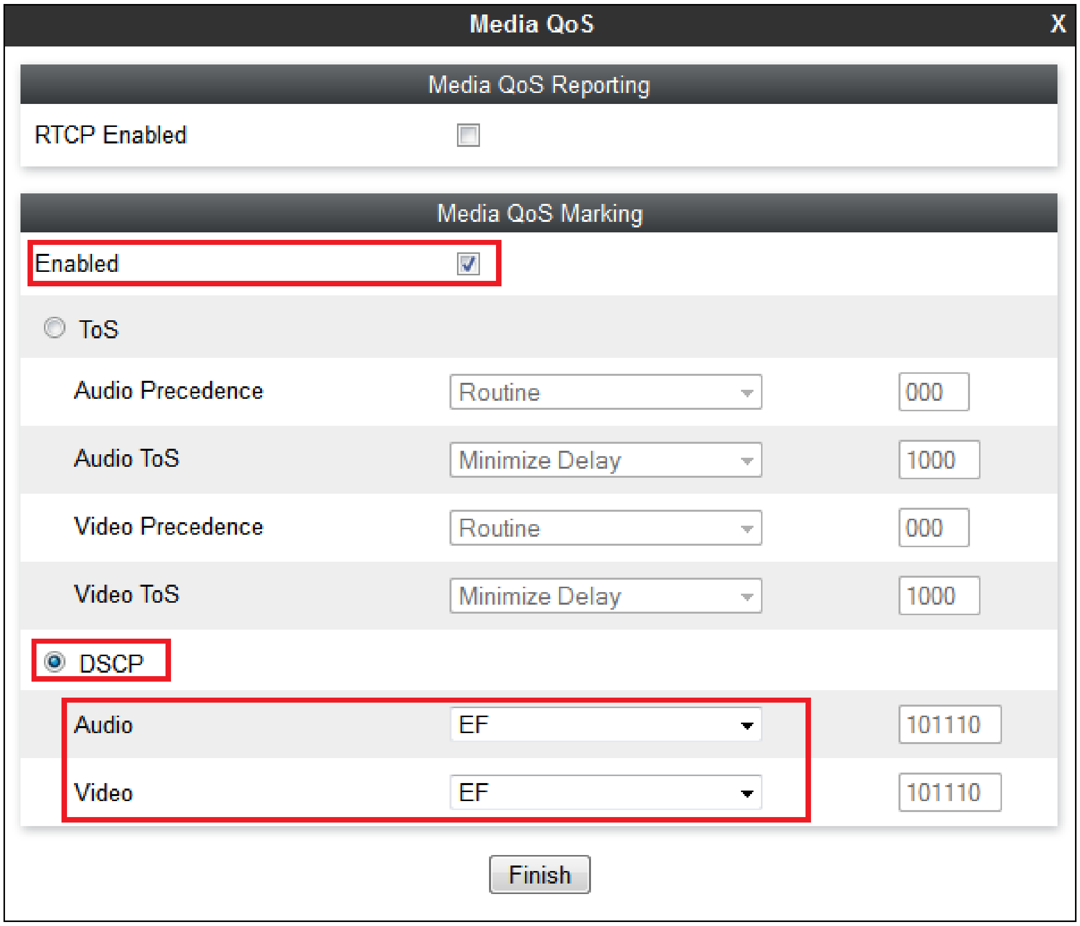


On the Avaya SBCE, Media Silencing feature detects silence when the call is in progress. If silence is detected and exceeds the allowed duration, the ASBCE generates an alert in the **Incidents Log**. In the compliance testing, the Media Silencing detection was disabled to prevent the call from unexpectedly disconnecting due to a RTP packet lost on the public Internet.

To modify Media Silencing, select the **Media Silencing** tab and click on the **Edit** button (not shown). Then uncheck **Media Silencing** and click on the **Finish** button.



Under **Media QoS** tab, click on the **Edit** button (not shown) to configure the Quality of Service (QoS). The ASBCE can be configured to mark the Differentiated Services Code Point (DSCP) in the IP packet header with specific values to support Quality of Services policies for the media. The **Audio** and **Video** are set to **EF** as recommended by service provider. The following screen shows the QoS values used for the compliance testing.



7.3.3. Signaling Rules

Signaling Rules define actions 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 ASBCE, 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.

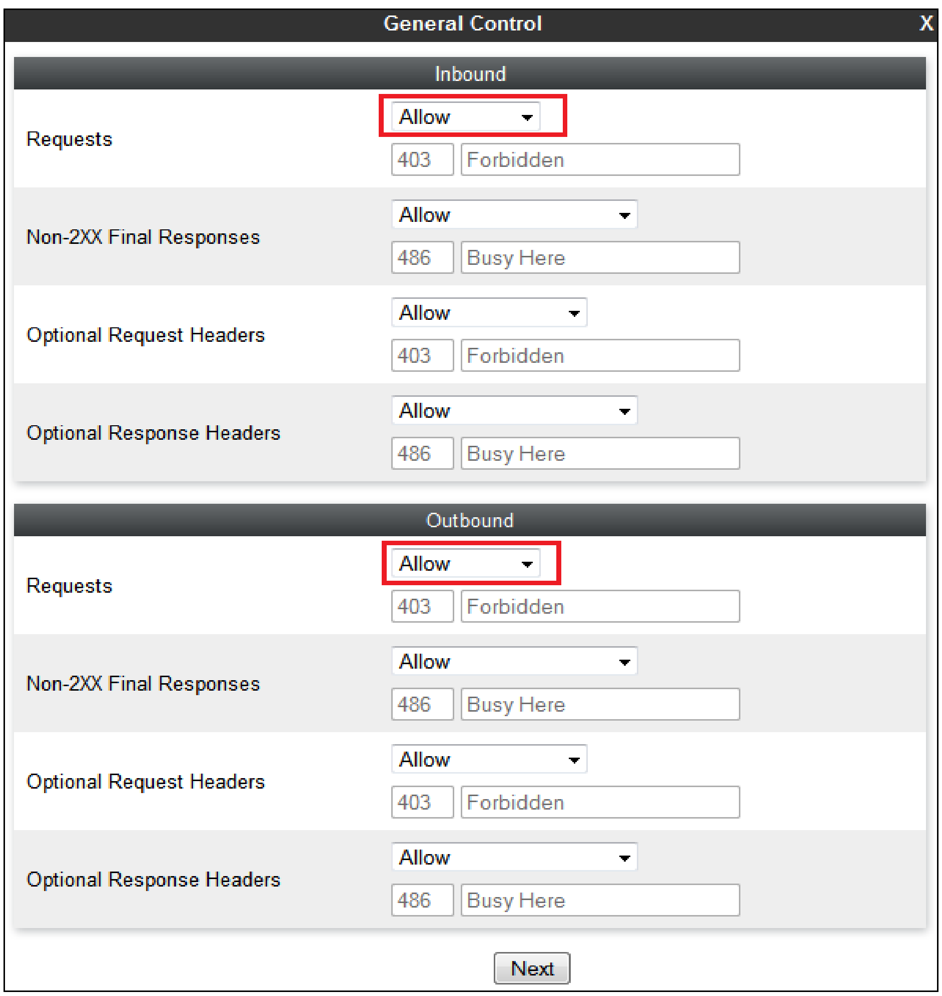
To clone a signaling rule, navigate to **Domain Policies → Signaling Rules**, select the **default** rule then click on the **Clone** button (not shown).

In the compliance testing, two **Signaling Rules** were created for Windstream and the CS1000.

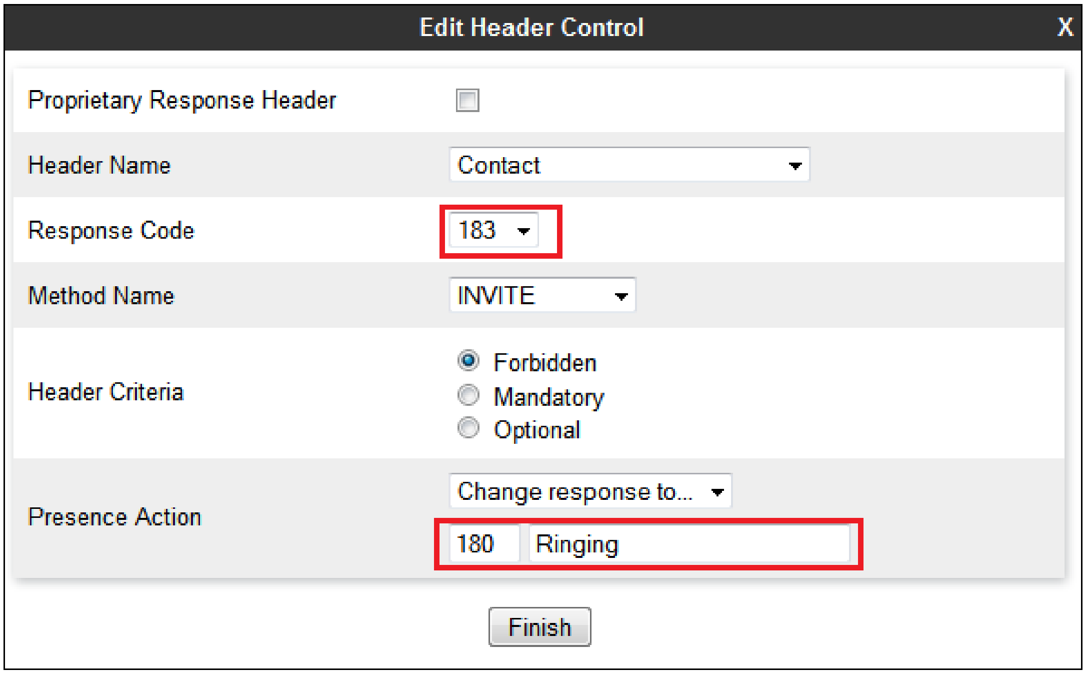
7.3.3.1 Signaling Rule for Windstream

Clone a Signaling Rule with a descriptive name e.g. **SigR_Windstream** and click on the **Finish** button.

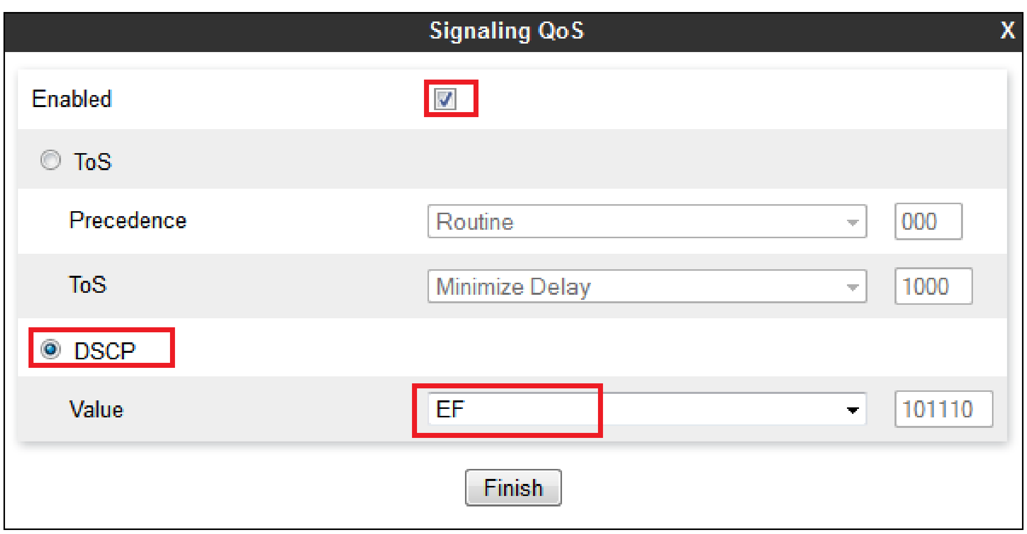
The **SigR_Windstream** was configured to allow the ASBCE to accept inbound and outbound call requests from Windstream. Cloning the Signaling Rule default, the **SigR_Windstream** will block all requests with a "403 Forbidden". To start accepting calls, go to **General Control** tab, click on the **Edit** button (not shown). Then change **Inbound** and **Outbound Requests** to **Allow** as shown in following screenshot.



On the **Response Headers** tab, click on **Add In Header Control** button to add a header control. This header will modify “183 SDP Session Progress” message responded from Windstream to “180 Ringing”. This header control combines with Server Interworking in **Section 7.4.2.2** to fix the ring back tone issue for the call blind transfer.

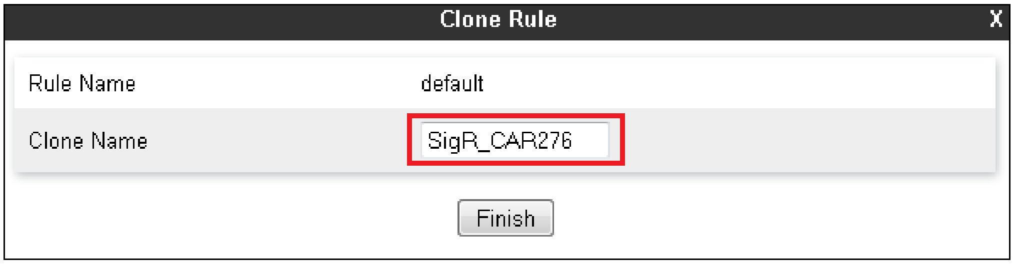


On the **Signaling QoS** tab, select the proper Quality of Service (QoS). The ASBCE can be configured to mark the **Differentiated Services Code Point (DSCP)** in the IP packet header with specific values to support Quality of Services policies for signaling. The following screen shows the QoS value used for the compliance testing.

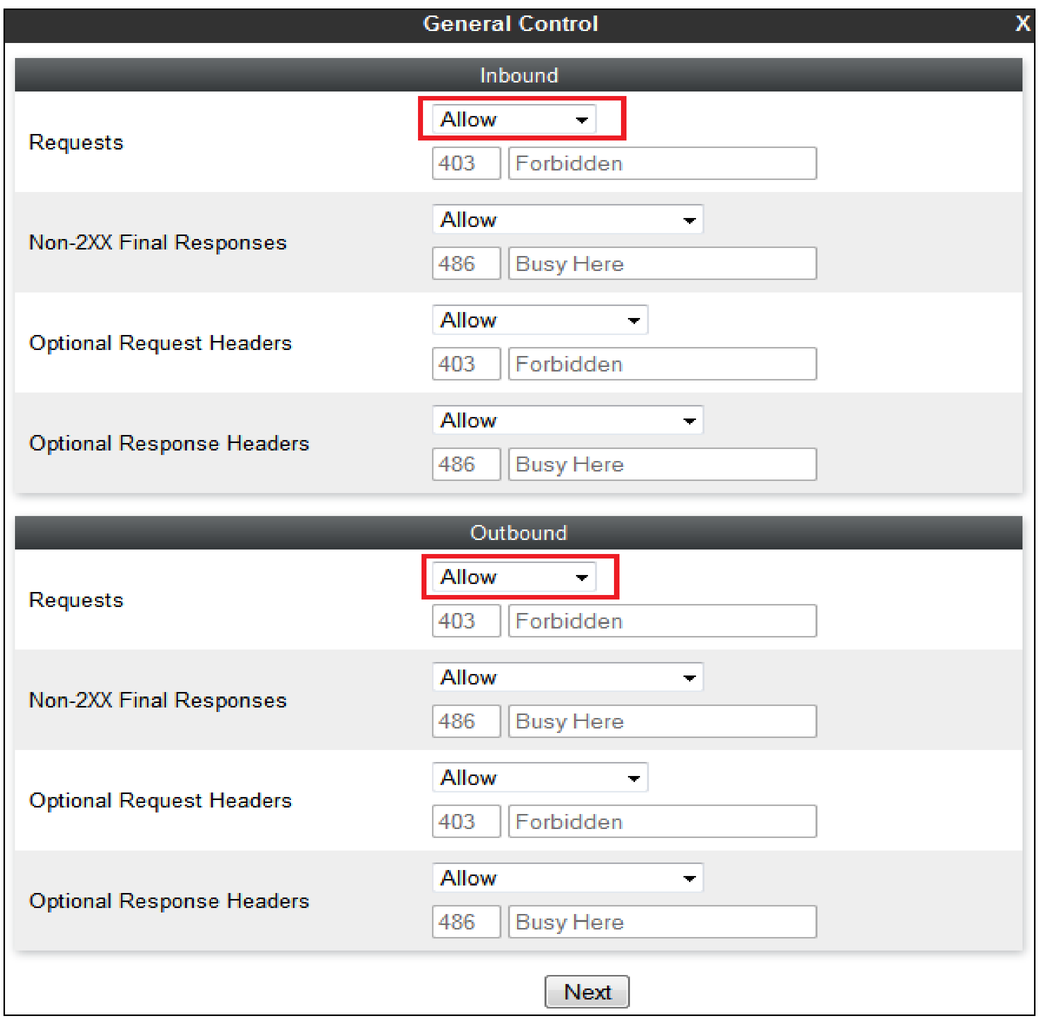


7.3.3.2 Signaling Rule for the CS1000

Clone a Signaling Rule with a descriptive name e.g. **SigR_CAR276** for the CS1000 and click on the **Finish** button.



This **SigR_CAR276** is configured to allow the ASBCE to accept inbound and outbound call requests from the CS1000. Cloning the **S**ignaling Rule **default**, the **SigR_CAR276** will block all requests with a "403 Forbidden". To start accepting calls, select **SigR_CAR276** and then go to **General** tab. Click on the **Edit** button (not shown). Then change **Inbound-Requests** and **Outbound-Requests** to **Allow** as shown in the following screenshot.



On the **Signaling QoS** tab, select the proper Quality of Service (QoS). The ASBCE can be configured to mark the **Differentiated Services Code Point (DSCP)** in the IP packet header with specific values to support Quality of Services policies for signaling. The following screen shows the QoS value used for the compliance testing.

7.3.4. Endpoint Policy Groups

The rules created within the Domain Policy section are assigned to an Endpoint Policy Group. The Endpoint Policy Group is then applied to a Server Flow defined in the next section**.**

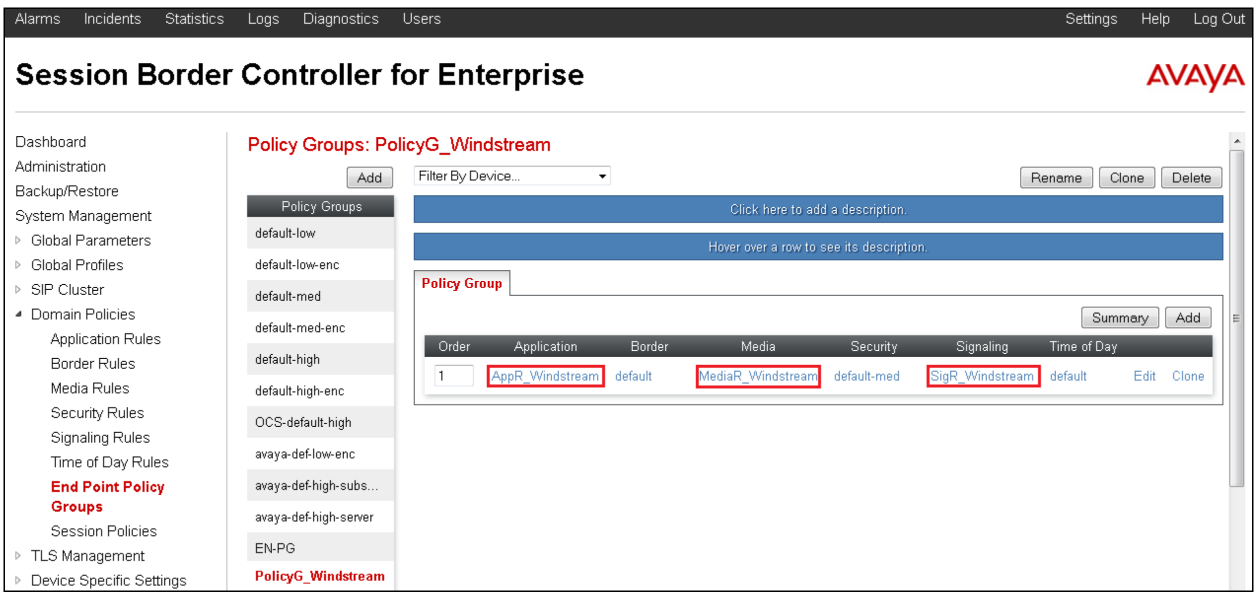
Two Endpoint Policy Groups were separately created for Windstream and the CS1000.

To create a policy group, navigate to **System Management → Domain Policies → Endpoint Policy Groups** and click on the **Add** button.

7.3.4.1 Endpoint Policy Group for Windstream

The following screen shows **PolicyG_Windstream** created for Windstream.

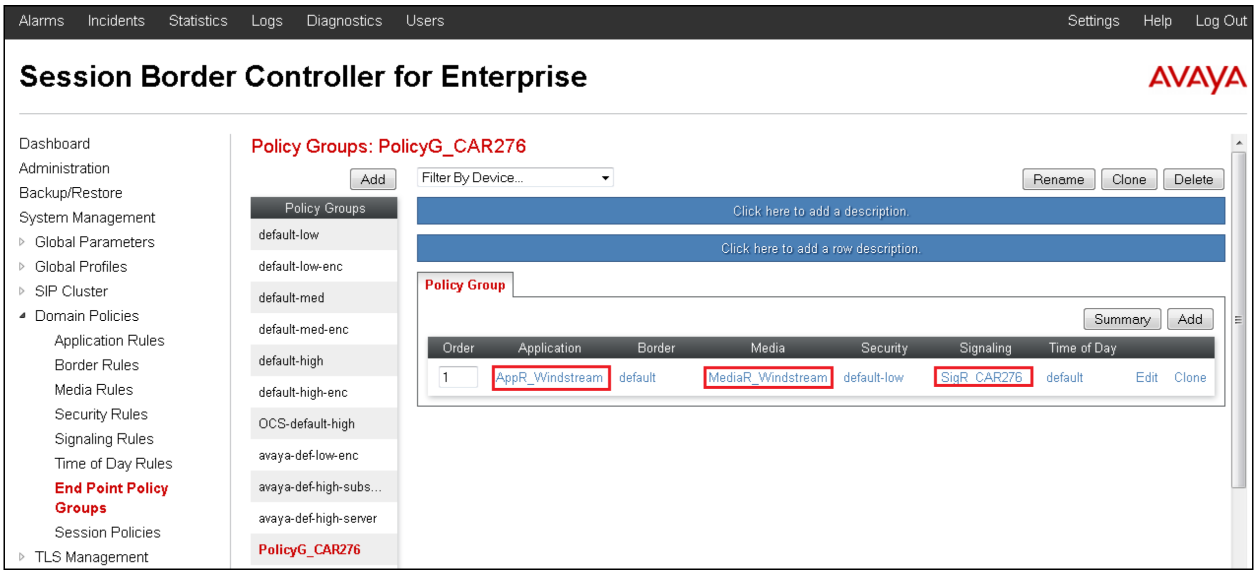
- Set **Application** Rrule to **AppR_Windstream** which was created in **Section 7.3.1**.
- Set **Media** rule to **MediaR_Windstream** which was created in **Section 7.3.2**.
- Set **Signaling** rule to **SigR_Windstream** which was created in **Section 7.3.3.1**.
- Set **Border** and **Time of Day** rules to **default**.
- Set **Security** rule to **default-med**.



7.3.4.2 Endpoint Policy Group for the CS1000

The following screen shows policy group **PolicyG_CAR276** created for the CS1000.

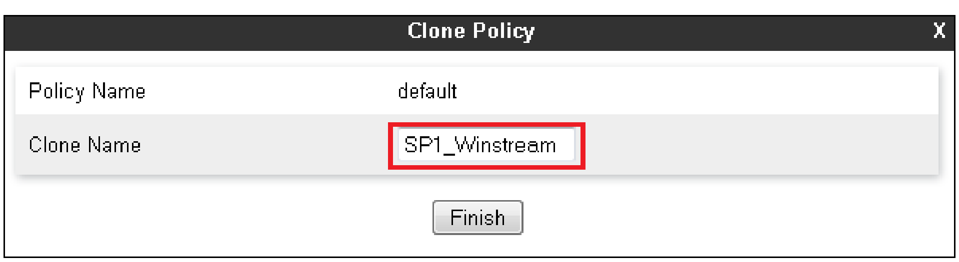
- Set **Application** rule to **AppR_Windstream** which was created in **Section 7.3.1**.
- Set **Media** rule to **MediaR_Windstream** which was created in **Section 7.3.2**.
- Set **Signaling** rule **SigR_CAR276** which was created in **Section 7.3.3.2**.
- Set the **Border** and **Time of Day** rules to **default**.
- Set the **Security** rule to **default-low**.



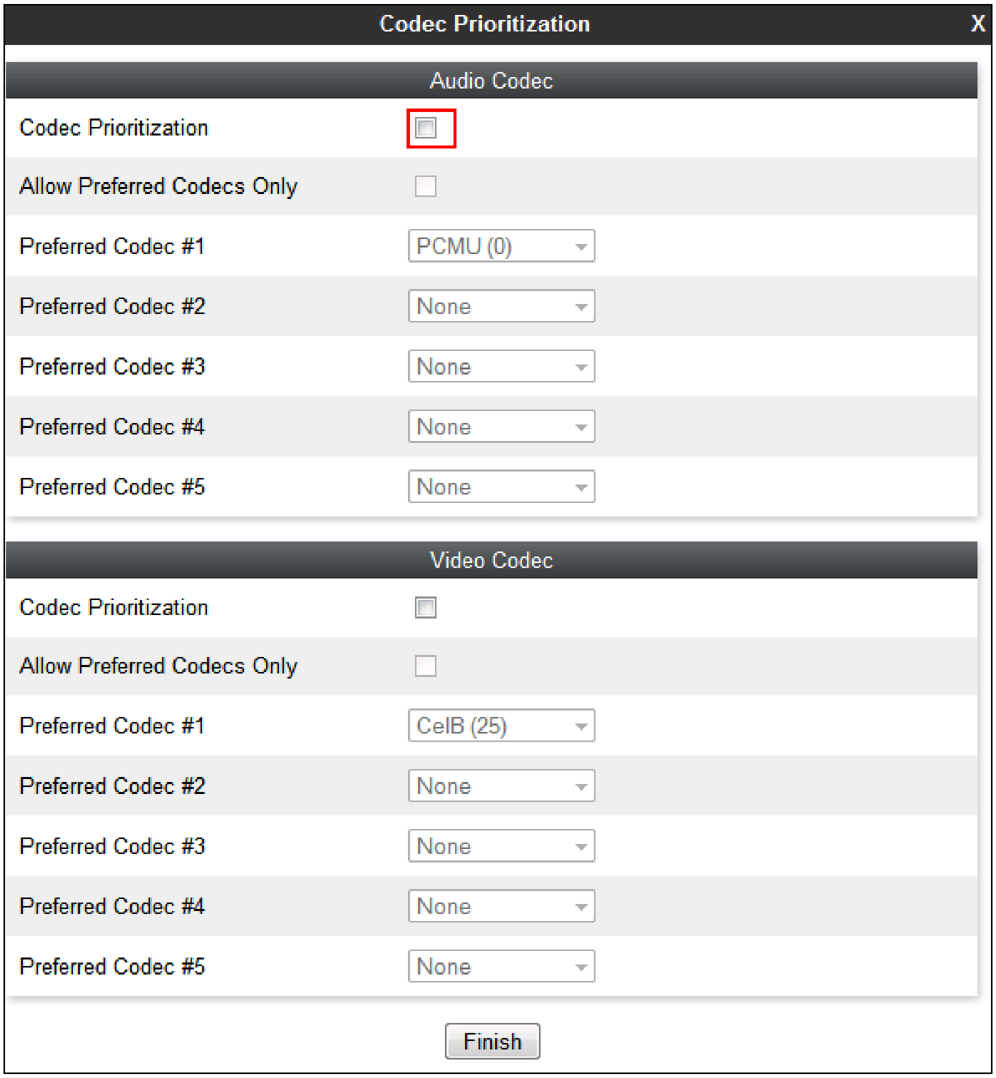
7.3.5. Session Policy

Session Policy is applied based on the source and destination of a media session i.e., which codec is to be applied to the media session between its source and destination. The source and destination are defined in URI Group in **Section 7.2.1**.

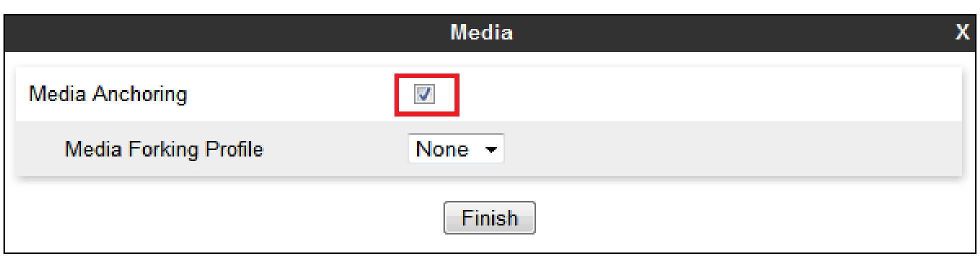
In the compliance testing, the Session Policy **SP1_Windstream** was created to match the codec configuration on Windstream. The policy also allows the ASBCE to anchor media in off-net call forward and call transfer scenarios. To clone a common Session Policy which applies to both Windstream and the CS1000, navigate to **Domain Policies → Session Policies**, select the **default** rule then click on the **Clone** button (not shown). Enter a descriptive name, .e.g. **SP1_Windstream** for the new policy and click on the **Finish** button.



Windstream supports voice codec G.711MU and G.729. To define **Codec Prioritization** for Audio Codecs, select the profile **SP1_Windstream** created above, click on the **Edit** button (not shown) and leave the **Codec Prioritization** option unchecked. With this configuration, ASBCE will pass all codecs that are supported and sent by CS1000 to Windstream and vice versa.



Under **Media** tab of the Session Policy **SP1_Windstream** created above, click on the **Edit** button (not shown). Check on **Media Anchoring** to allow the Avaya SBCE to anchor media in off-net call forward and call transfer scenarios.



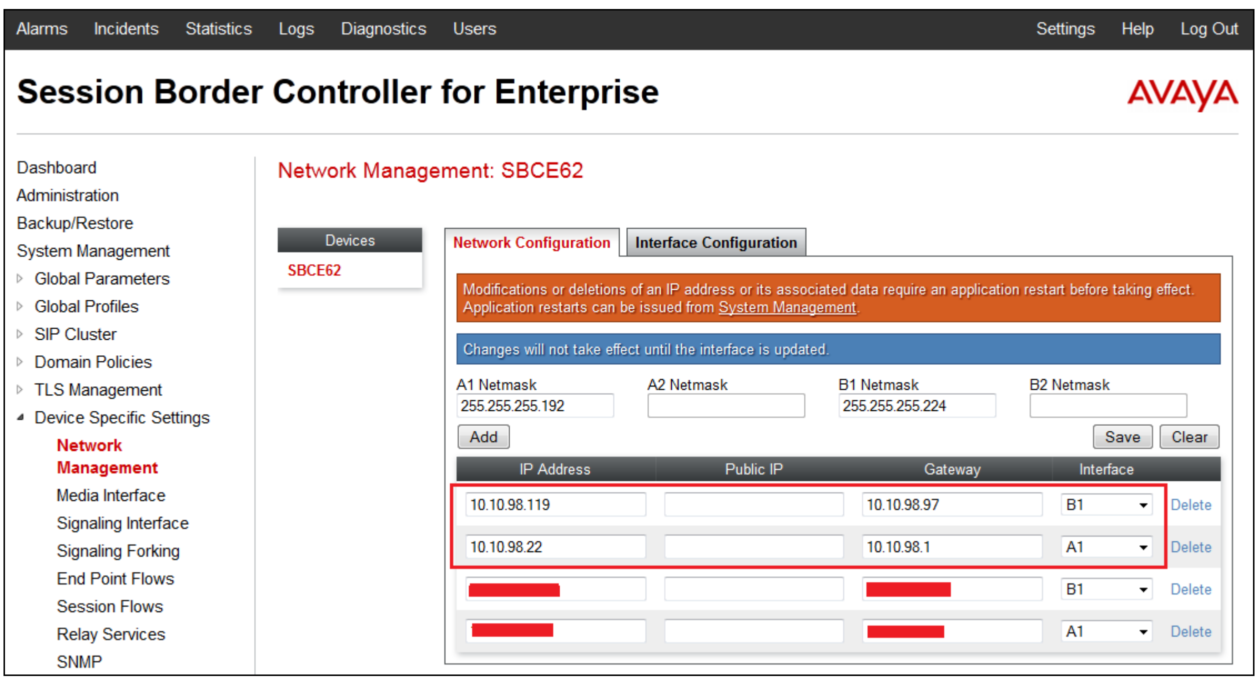
7.4. Device Specific Settings

Device Specific Settings feature allows aggregate system information to be viewed and various device-specific parameters to be managed to determine how a particular device will function when deployed in the network. Specifically, it gives the ability to define and administer various device-specific protection features such as Message Sequence Analysis (MSA) functionality and protocol scrubber rules, end-point and session call flows, as well as the ability to manage system logs and control security features.

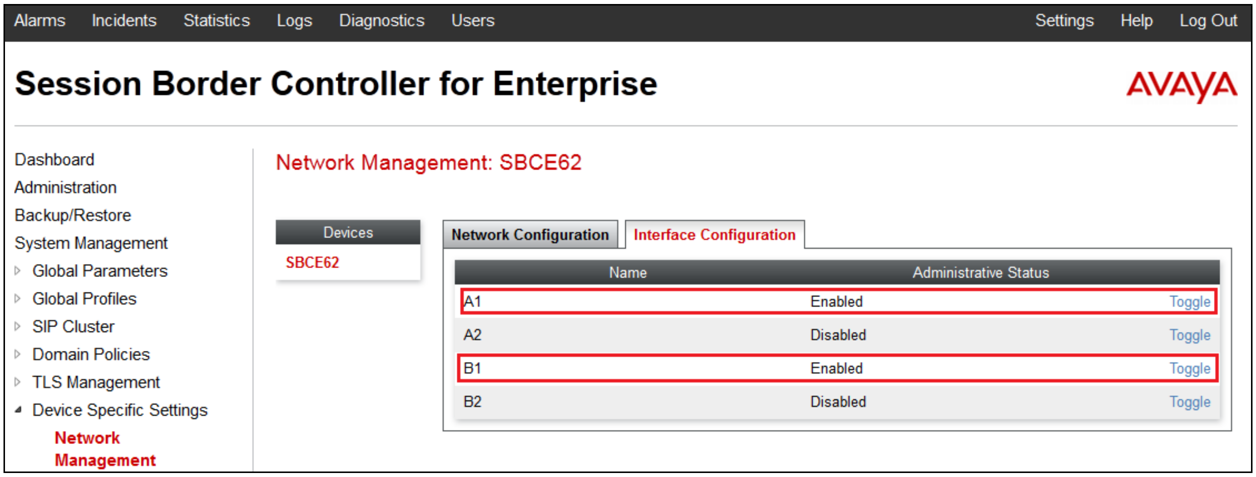
7.4.1. Network Management

Network Management page is where the network interface settings are configured and enabled. During the installation process of the Avaya SBCE, certain network-specific information is defined such as device IP addresses, public IP addresses, subnet mask, gateway, etc. to interface the device to the network. This information populates the various Network Management tabs, which can be edited as needed to optimize device performance and network efficiency.

Navigate to **System Management → Device Specific Settings → Network Management**. Under **Network Configuration** tab, verify the IP addresses assigned to the interfaces and that the interfaces are enabled. The following screen shows the private interface is assigned to **A1** and the public interface is assigned to **B1** appropriate to the parameters shown in **Figure 1**.



On the **Interface Configuration** tab, enable the interfaces connecting to the inside and outside networks. To enable an interface, click it's **Toggle State** button. The following screen shows interface **A1** and **B1** were **Enabled**.



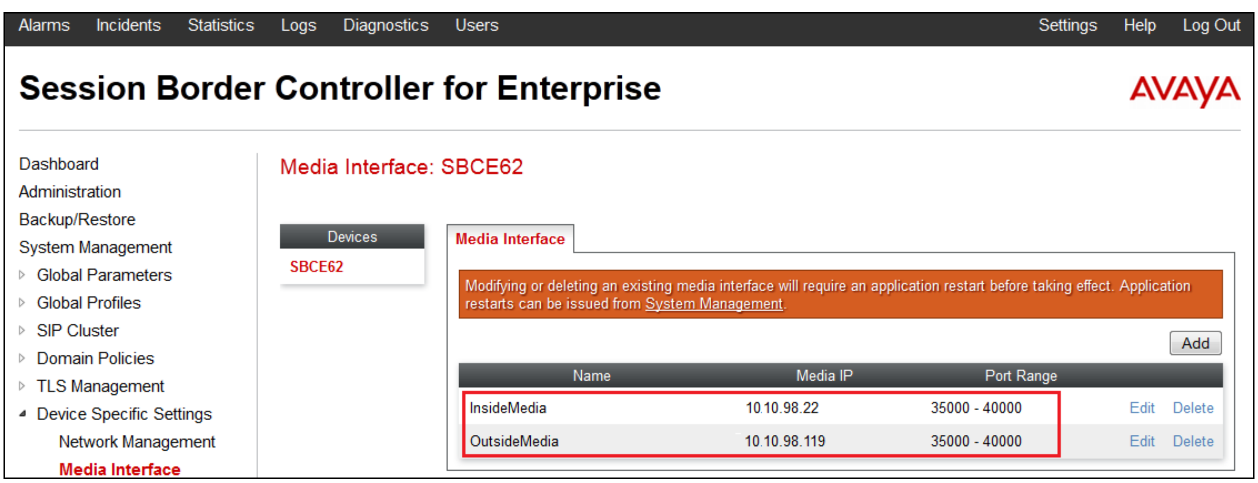
7.4.2. Media Interface

Media Interface screen is where the media ports are defined. The ASBCE will open connections for RTP traffic on the defined ports.

To create a new **Media Interface**, navigate to **System Management → Device Specific Settings → Media Interface** and click on the **Add** button.

Two separate Media Interfaces are needed for both the inside and outside interfaces. The following screen shows the Media Interfaces **InsideMedia** and **OutsideMedia** were created for the compliance testing.

Note: After the media interfaces are created, an application restart is necessary before the changes will take effect.

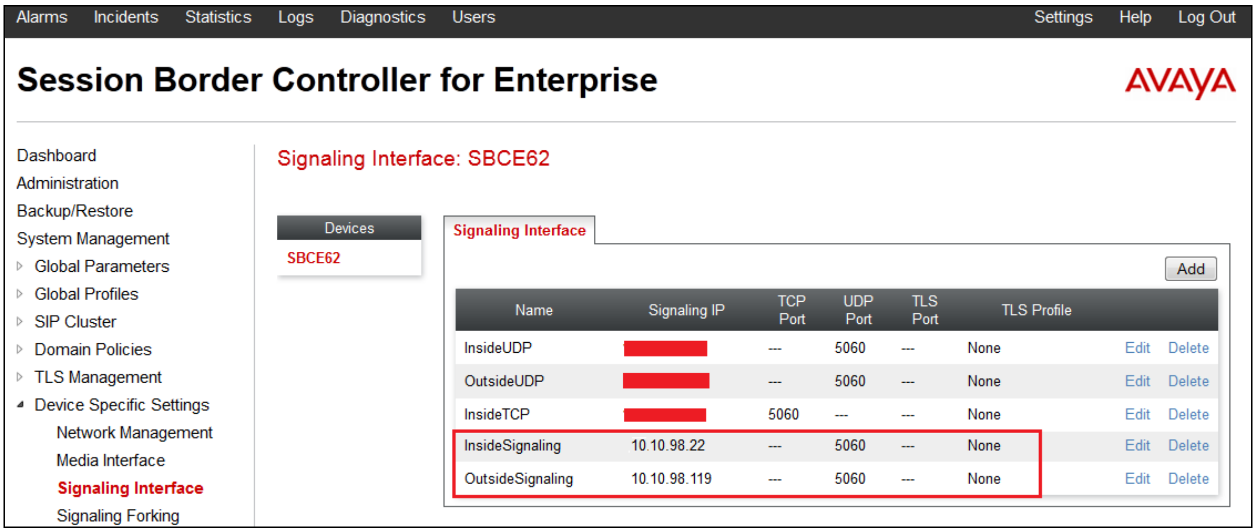


7.4.3. Signaling Interface

Signaling Interface screen is where the SIP signaling port is defined. The ASBCE will listen for SIP requests on the defined port.

To create a new **Signaling Interface**, navigate to **System Management → Device Specific Settings → Signaling Interface** and click on the **Add Signaling Interface** button.

Two separate Signaling Interfaces are needed for both inside and outside interfaces. The following screen shows the Signaling Interfaces **InsideSignaling** and **OutsideSignaling** were created in the compliance testing with **UDP/5060** for both inside and outside interfaces.



7.4.4. End Point Flows - Server Flow

When a packet is received by ASBCE, the content of the packet (IP addresses, URIs, etc.) is used to determine which flow it matches. Once the flow is determined, the flow points to a policy which contains several rules concerning processing, privileges, authentication, routing, etc. Once routing is applied and the destination endpoint is determined, the policies for this destination endpoint are applied. The context is maintained, so as to be applied to future packets in the same flow.

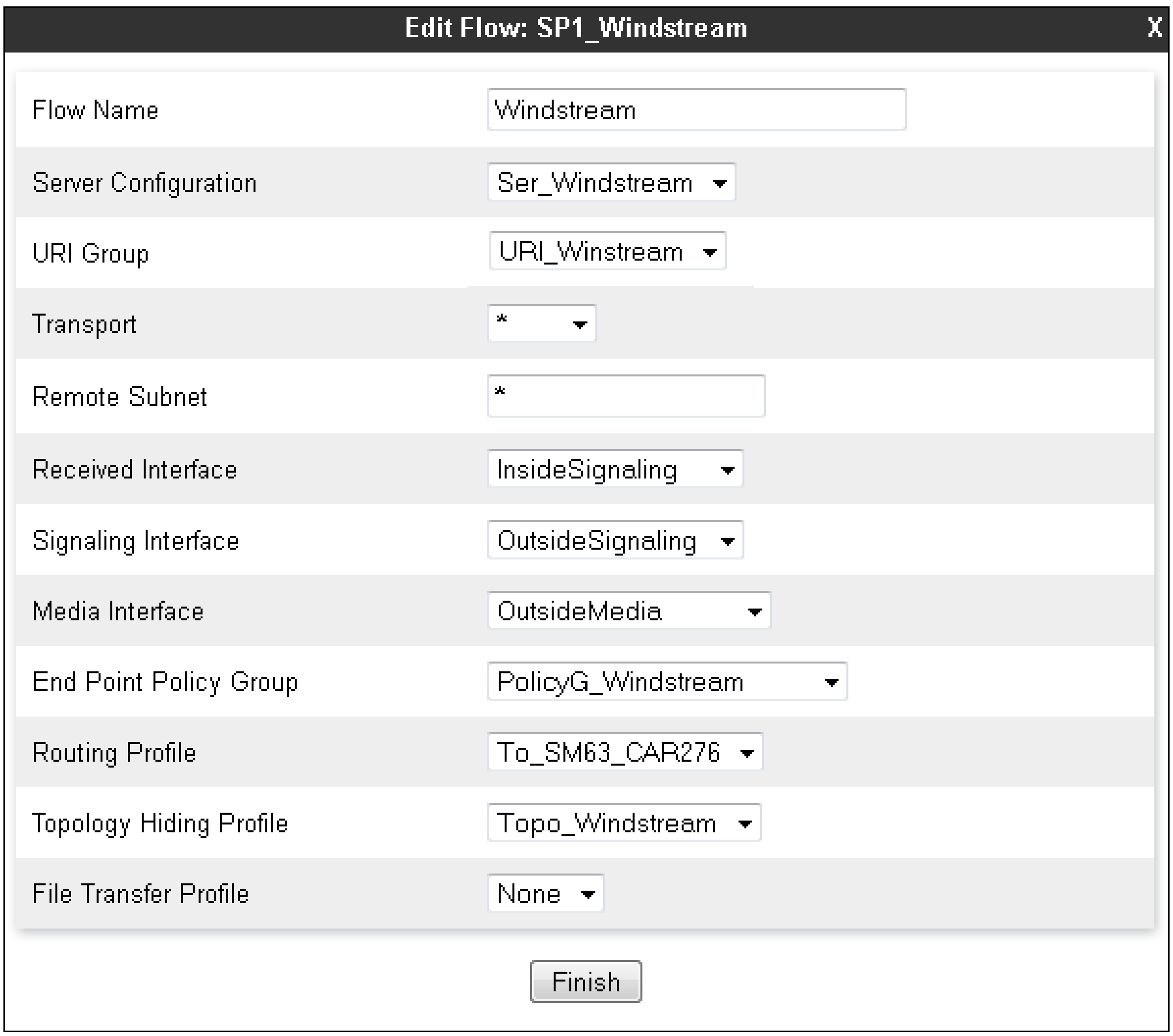
In the compliance testing, two separate Server Flows were created for Windstream and Session Manager.

To create a Server Flow, navigate to **System Management → Device Specific Settings → End Point Flows**, select the **Server Flows** tab and click on the **Add** button (not shown). In the new window that appears, enter the following values while the other fields are kept as default.

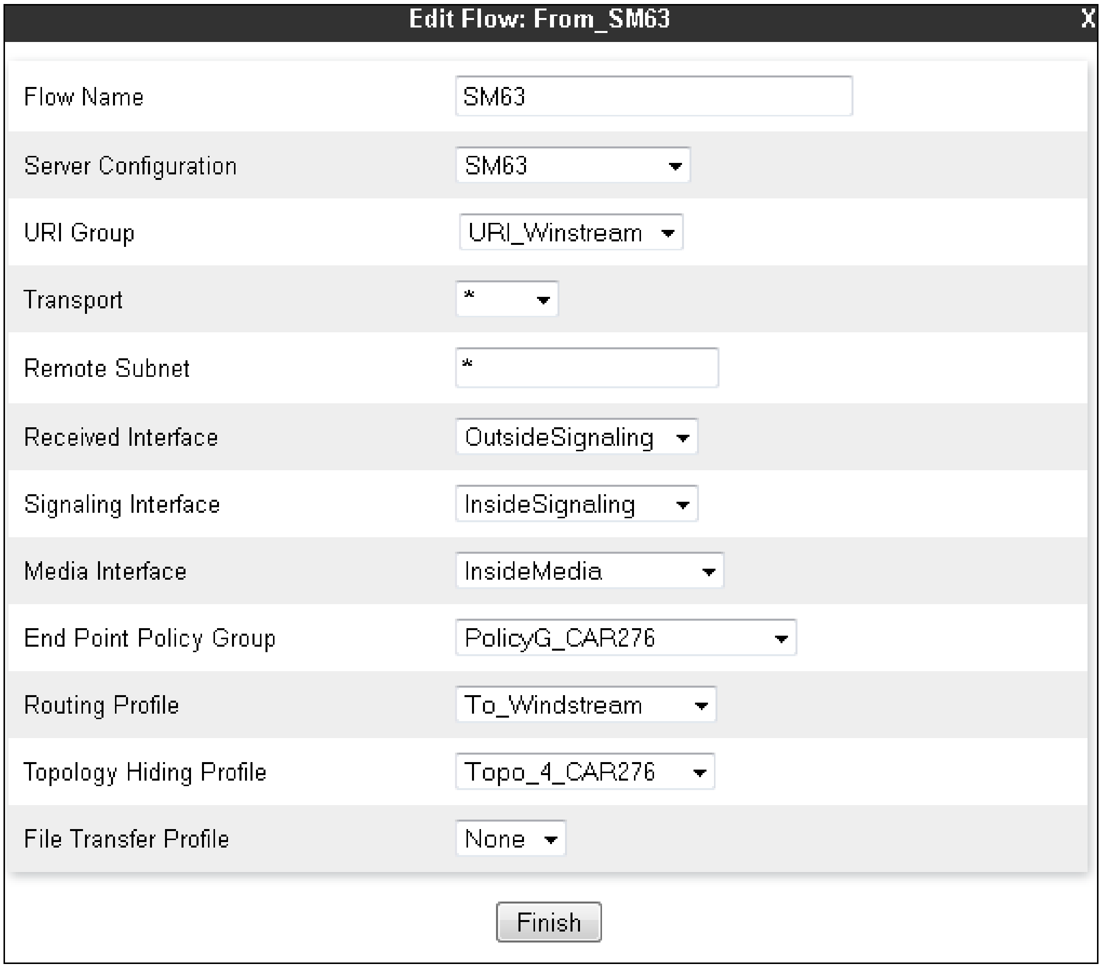
- **Flow Name**: Enter a descriptive name.
- **Server Configuration**: Select the Server Configuration **Ser_Windstream** created in **Section 0.1.**
- **URI Group**: Select the URI Group **URI_Windstream** created in **Section 7.2.1**.

- **Received Interface**: Select the Signaling Interface created in **Section 7.4.3** which the Server Configuration will use for receiving SIP signaling.
- **Signaling Interface**: Select the Signaling Interface created in **Section 7.4.3** which the Server Configuration will use for sending SIP signaling.
- **Media Interface**: Select the Media Interface created in **Section 7.4.2** which the Server Configuration will use for sending RTP..
- **End Point Policy Group**: Select the End Point Policy Group created in **Section 7.3.4**.
- **Routing Profile**: Select the Routing Profile created in **Section 7.2.2** which the Server Configuration will use for routing calls.
- **Topology Hiding Profile**: Select the Topology Hiding profile created in **Section 7.2.3** to apply toward the Server Configuration.
- Use default values for all remaining fields. Click **Finish** to save and exit.

The following screen shows the Server Flow named **Windstream** for Windstream.



The following screen shows the Server Flow named **SM63** for Session Manager.



7.4.5. Session Flows

Session Flows feature allows defining certain parameters that pertain to the media portions of a call, whether it originates from the enterprise or outside the enterprise. This feature provides the complete and unparalleled flexibility to monitor, identify and control very specific types of calls based upon these user-definable parameters. Session Flows profiles SDP media parameters, to completely identify and characterize a call placed through the network.

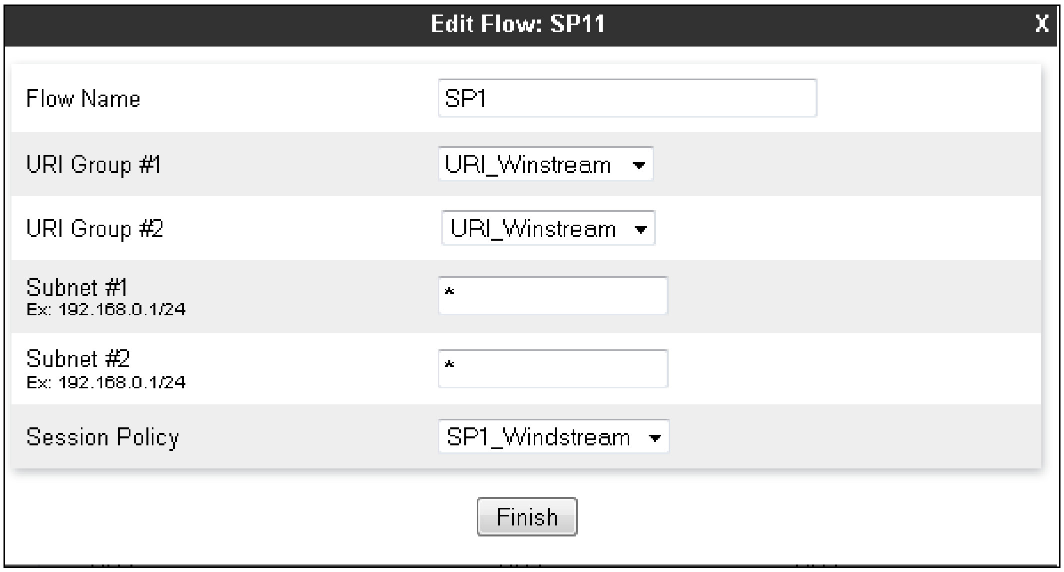
A common Session Flow **SP1** was created for both the Windstream and the CS1000.

To create a session flow, navigate to **System Management → Device Specific Settings → Session Flows** then click on the **Add Flow** button (not shown). In the new window that appears, enter the following values while the remaining fields are kept as default.

- **Flow Name**: Enter a descriptive name.
- **URI Group #1**: Select the URI Group created in **Section 7.2.1** to assign to the Session Flow as the source URI Group.

- **URI Group #2**: Select the URI Group created in **Section 7.2.1** to assign to the Session Flow as the destination URI Group.
- **Session Policy**: Select the Session Policy created in **Section 7.3.5** to assign to the Session Flow.
- Click on the **Finish** button.

The following screen shows the Session Flow named **SP1**.



8. Configure Windstream SIP Trunking Service

Windstream is responsible for the configuration of its SIP Trunking Service. The customer will need to provide the IP address used to reach the ASBCE at enterprise side. Windstream will provide the customer with the necessary information to configure the SIP Trunk connection from enterprise to the Windstream.

The information provided by Windstream includes:

- IP address of the Windstream SIP proxy.
- Service provider public SIP domains.
- Supported codecs.
- DID numbers.
- IP addresses and port numbers used for signaling or media through any security devices.
- A customer specific SIP signaling reference.

9. Verification

This section provides verification steps that may be performed in the field to verify that the solution is configured properly.

9.1. Verification Steps

The following activities are made to each test scenario.

- Calls are checked for the correct call progress tones and cadences.
- During the ringing state, the ring back tone and destination ringing are checked.
- Calls are checked in both hands-free and handset mode due to internal Avaya requirement.
- Calls are checked for speech path in both directions using spoken words to ensure clarity of speech.
- The display(s) of the sets/clients involved are checked for consistent and expected calling party name and number and redirection information both prior to answer and after call establishment.
- The speech path and messaging system are observed for timely and quality End to End tone audio path generation and application responses.
- The call server maintenance terminal window is used for the monitoring of BUG(s), ERR and AUD messages.
- Speech path and display checked before and after calls are put on/off hold from each end.
- Applicable files are screened on an hourly basis during the testing for messages that may indicate technical issues. This refers to Avaya PBX files.
- Calls are checked to ensure that all resources such as Virtual trunks, TDM trunks, Sets and VGWs are released when a call scenario ends.

9.2. Protocol Traces

The following SIP message headers are inspected using sniffer traces:

- Request-URI: Verify the request number and SIP domain.
- From: Verify the display name and display number.
- To: Verify the display name and display number.
- P-Asserted-Identity: Verify the display name and display number.
- Privacy: Verify privacy masking with “user, id”.
- Diversion: Verify DID number.
- Authorization: Verify Digest Authentication implementation.

The following attributes in SIP message body are inspected using sniffer traces:

- Connection Information (c line): Verify IP addresses of near and far endpoints.
- Time Description (t line): Verify session timeout value of near and far endpoints.
- Media Description (m line): Verify audio port, codec, DTMF event description.
- Media Attribute (a line): Verify specific audio port, codec, ptime, send/ receive abilities, DTMF event and fax attributes.

a) **SIP/INVITE from CS1000 captured at Avaya SBCE OUTSIDE interface.**

```
INVITE sip:6139675258@64.xxx.xx.174;user=phone SIP/2.0
From: "OfficeE 1490" sip:501xxx1490@10.10.98.119;user=phone
;tag=34d3d50-aa610a87-13c4-55013-54cba-321577fb-54cba
To: <sip:6139675258@64.xxx.xx.174;user=phone>
CSeq: 1 INVITE
Call-ID: fa9d8b43e2953717f450c54e0f127b40
Contact: <sip:501xxx1490@10.10.98.119:5060;transport=udp;user=phone;
gsid=f60d0850-9e48-11e3-87c0-e41f13b32ca8>
Record-Route: <sip:10.10.98.119:5060;ipcs-line=340148;lr;transport=udp>
Allow: INVITE, ACK, BYE, REGISTER, REFER, NOTIFY, CANCEL, PRACK, OPTIONS,
INFO, SUBSCRIBE, UPDATE
Supported: 100rel, x-nortel-sipvc, replaces
User-Agent: Nortel CS1000 SIP GW release_7.0 version_ssLinux-7.65.16 AVAYA-SM-
6.3.4.0.634014
Max-Forwards: 30
Via: SIP/2.0/UDP 135.10.98.119:5060;branch=z9hG4bK-s1632-000536499325-1--s1632-
Alert-Info: <cid:external@avayalab.com>
Privacy: none
P-Asserted-Identity: "OfficeE 1490" <sip:501xxx1490@10.10.98.119;user=phone>
History-Info: <sip:6139675258@64.xxx.xx.174;user=phone>;index=1
History-Info:<sip:6139675258@avayalab.com;user=phone>;index=1.1
Remote-Address: 10.33.5.6:5201:1:2
Content-Type: application/sdp
P-AV-Message-Id: 1_1
x-nt-e164-clid: +5012871490@avayalab.com;user=phone
P-Charging-Vector: icid-value="f60d0850-9e48-11e3-87c0-e41f13b32ca8"
Av-Global-Session-ID: f60d0850-9e48-11e3-87c0-e41f13b32ca8
P-Location:
SM;origlocname="Belleville";origsiglocname="Belleville";origmedialocname="Belleville";
termlocname="Belleville";termsiglocname="Belleville";smaccounting="true"
Content-Length: 264
v=0
o=- 257 1 IN IP4 10.10.98.119
s=-
c=IN IP4 10.10.98.119
t=0 0
m=audio 35464 RTP/AVP 0 8 18 101 111
c=IN IP4 135.10.98.119
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtpmap:111 X-nt-inforeq/8000
a=ptime:20
a=sendrecv
```

b) SIP/183 SDP responded from Windstream

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 10.10.98.119:5060;branch=z9hG4bK-s1632-000536499325-1--s1632-
Record-Route: <sip:10.10.98.119:5060;ipcs-line=340148;lr;transport=udp>
From: "OfficeE 866972" <sip:5012871490@010.10.98.119;user=phone>;tag=34d3d50-
aa610a87-13c4-55013-54cba-321577fb-54cba
To:
<sip:6139675258@64.xxx.xx.174;user=phone>;tag=fe6eU1vW9o3YmqAsCF9CEFCB62614da6
Call-ID: fa9d8b43e2953717f450c54e0f127b40
CSeq: 1 INVITE
Contact: <sip:6139675258@64.xxx.xx.174:5060>
Content-Type: application/sdp
Content-Length:   177
v=0
o=- 1782959037 1782959037 IN IP4 64.xxx.xx.174
s=-
c=IN IP4 64.xxx.xx.174
t=0 0
m=audio 16604 RTP/AVP 0 101
a=rtpmap:101 telephone-event/8000
a=ptime:20
a=sendrecv
```

c) SIP/200 SDP responded from Windstream

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.10.98.119:5060;branch=z9hG4bK-s1632-000536499325-1--s1632-
Record-Route: <sip:10.10.98.119:5060;ipcs-line=340148;lr;transport=udp>
From: "OfficeE 1490" <sip:5012871490@10.10.98.119;user=phone>;tag=34d3d50-aa610a87-
13c4-55013-54cba-321577fb-54cba
To:
<sip:6139675258@64.xxx.xx.174;user=phone>;tag=fe6eU1vW9o3YmqAsCF9CEFCB62614da6
Call-ID: fa9d8b43e2953717f450c54e0f127b40
CSeq: 1 INVITE
Contact: <sip:6139675258@64.xxx.xx.174:5060>
Content-Type: application/sdp
Content-Length:   177
v=0
o=- 1782959037 1782959037 IN IP4 64.xxx.xx.174
s=-
c=IN IP4 64.xxx.xx.174
t=0 0
m=audio 16604 RTP/AVP 0 101
a=rtpmap:101 telephone-event/8000
a=ptime:20
a=sendrecv
```

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Communication Server 1000 Release 7.6, Avaya Aura® Session Manager Release 6.3 and the Avaya Session Border Controller for Enterprise Release 6.2 to Windstream SIP Trunking Service. Windstream SIP Trunking Service is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises.

All of the test cases have been executed. Despite the number of observations and limitations seen during testing as noted in **Section 2.2**, the test results met the objectives outlined in **Section 2.1**. The Windstream SIP Trunking Service is considered compliant with Avaya Communication Server 1000 Release 7.6, Avaya Aura® Session Manager Release 6.3 and the Avaya Session Border Controller for Enterprise Release 6.2.

11. References

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

[1] *Network Routing Service Fundamentals, Avaya Communication Server 1000,* Release 7.6, Document Number NN43001-130, Revision 03.02, Jun 2013.
[2] *IP Peer Networking Installation and Commissioning, Avaya Communication Server 1000,* Release 7.6, Document Number NN43001-313, Revision: 05.02, Jun 2013.
[3] *Communication Server 1000E Overview, Avaya Communication Server 1000,* Release 7.6, Document Number NN43041-110, Revision: 05.02, Jun 2013.
[4] *Communication Server 1000 Unified Communications Management Common Services Fundamentals, Avaya Communication Server 1000,* Release 7.6, Document Number NN43001-116, Revision 05.08, Jun 2013.
[5] *Communication Server 1000 Dialing Plans Reference, Avaya Communication Server 1000,* Release 7.5, Document Number NN43001-283, Revision 05.02, November 2010.
[6] *Product Compatibility Reference, Avaya Communication Server 1000,* Release 7.6, Document Number NN43001-256, Revision 05.02, Jun 2013.
[7] *Installing and Configuring Avaya Aura® System Platform*, Release 6.3.1, Oct 2013.
[8] *Administering Avaya Aura® System Platform*, Release 6.3.1, Oct 2013.
[9] *Installing and Upgrading Avaya Aura® System Manager*, Release 6.3, Oct 2013.
[10] *Maintaining and Troubleshooting Avaya Aura® Session Manager*, Release 6.3, Oct 2013, Document Number 03-603473.
[11] *Administering Avaya Aura® Session Manager,* Release 6.3, Oct 2013, Document Number 03-603324.
[12] *Administering Avaya Session Border Controller for Enterprise*, Release 6.2, Issue 2, Jan 2013.
[13] *RFC3261 SIP: Session Initiation Protocol*, http://www.ietf.org/.
[14] *RFC3262, Reliability of Provisional Responses in the Session Initiation Protocol (SIP)* http://www.ietf.org/.
[15] *RFC2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals,* http://www.ietf.org/.

Product documentation for Windstream SIP Trunking Service is available from Windstream.

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.