Avaya Solution & Interoperability Test Lab

Application Notes for Configuring TELUS SIP Trunking with the Avaya Communication Server 1000 Release 7.5, Avaya Aura® Session Manager Release 6.2 and Avaya Session Border Controller for Enterprise Release 4.0.5 – Issue 1.0

Abstract

These Application Notes illustrate a sample configuration using Communication Server 1000 Release 7.5, Avaya Aura® Session Manager Release 6.2, Avaya Session Border Controller for Enterprise (Avaya SBCE) Release 4.0.5 with the TELUS system.

The TELUS offer referenced within these Application Notes is designed for business customers with an Avaya SIP trunk solution. The service provides local and/or long distance PSTN calling via standards-based SIP trunks directly, without the need for additional TDM enterprise gateways or TDM cards and the associated maintenance costs.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes illustrate a sample configuration using Communication Server 1000 Release 7.5, Avaya Aura® Session Manager Release 6.2, Avaya Session Border Controller for Enterprise (Avaya SBCE) Release 4.0.5 with the TELUS system. The TELUS Service provides local and/or long-distance calls (with PSTN endpoints) via standards-based SIP trunks.

2. General Test Approach and Test Results

The Communication Server 1000 connects to the Avaya SBCE via Session Manager using a SIP connection. Then the Avaya SBCE connects to the TELUS system using SIP signaling. Various call types were made from Communication Server 1000 to and from the TELUS system to verify the interoperability.

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

2.1. Interoperability Compliance Testing

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

- General call processing between Communication Server 1000 and TELUS systems including:
  - Codec/ptime (G.729/20ms, G.711 u-law/20ms)
  - Hold/Resume on both ends
  - CLID displayed
  - Ring-back tone
  - Speech path
  - Dialing plan support
  - Advanced features (Call on Mute, Call Park, Call Waiting)
  - Abandoned Call
- Call redirection verification: all supported methods (blind transfer, consultative transfer, call forward, and conference) including CLID. Call redirection is performed from both ends
- Fax with G.711
- DTMF in both directions
- SIP Transport UDP
- Thru dialing via the Communication Server 1000 Call Pilot
- Voice Mail Server Call Pilot (hosted on Avaya system)
- DV Endpoints

- TELUS Mobility Endpoints

The following assumptions were made for these compliance tested configuration:

1. Communication Server 1000 R7.5 software with latest patches
2. TELUS provides support to setup, configure and troubleshoot on carrier switch during testing execution.

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

1. Calls were checked for the correct call progress tones and cadences.
2. During the ringing state the ring back tone and destination ringing were checked.
3. Calls were checked in both hands-free and handset mode due to internal Avaya requirement.
4. Calls were checked for speech path in both directions using spoken words to ensure clarity of speech.
5. The display(s) of the sets/clients involved were checked for consistent and expected CLID and redirection information both prior to answer and after call establishment.
6. The speech path and messaging system were observed for timely and quality End to End tone audio path generation and application responses.
7. The call server maintenance terminal window was open during the test cases execution for the monitoring of BUG(s), ERROR and AUD messages.
8. Speech path was checked before and after calls were put on/off hold from each end.
9. Applicable files were screened on an hourly basis during the testing for message that may indicate technical issues. This refers to Communication Server files.
10. Calls were checked to ensure that all resources such as Virtual trunks, TDM trunks, Sets and VGWs are released when a call scenario ends.

2.2. Test Results

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

1. If the Communication Server 1000 phone holds/resume an outbound call, the dialed digits are no longer displayed. This is a Communication Server 1000 known issue.
2. PSTN1 phone calls to Communication Server 1000 phone, then phone performs a blind transfer to PSTN2 phone. PSTN1 phone could not hear ring-back-tone from PSTN2 phone when Communication Server 1000 phone completed the blind transfer. In this particular scenario, the UPDATE support is required on the Communication Server 1000, but the PSTN-to-SIP gateway that TELUS uses for this test case does not support the UPDATE. In order to make the blind transfer work, make sure to enable plug-in 501 on Communication Server 1000 to allow blind transfer to work without the UPDATE method. The limitation of this plug-in is that no ring-back-tone is provided to the originator of the call for the duration that the destination set is ringing.
3. Calls that are redirected on the Communication Server 1000 require a SIP Diversion header to be added so the calls can be handled properly on the TELUS network. The Diversion header is needed to fix billing situations within the TELUS network on the NSN HiQ where calls are forwarded or transferred to external sets. The NSN HiQ requires Diversion headers if the outgoing call contains a different number in the From

and PAI headers, which is the case on redirected calls. The Diversion header ensures that the proper party is billed for the call. The Communication Server 1000 does not support Diversion headers. In order to provide this functionality, the Avaya Aura ® Session Manager will extract the user and host information from the History-Info header and create a Diversion header (Refer to section 6.4.2 and 6.4.3).

4. The TELUS network does not support SIP History-Info headers as these headers are primarily used for inter-SIP PBX communication. Instead, the TELUS network requires that a SIP P-Asserted-Identity header be sent for redirected calls. The Communication Server 1000 accomplishes this by using the Avaya SBCE to extract the user and host information from the Diversion Header and create P-Asserted-Identity header (Refer to section 7.2.9).

It was agreed with TELUS that the above observations were not severe enough to fail the testing.

2.3. Support

For technical support on the Avaya products described in these Application Notes visit:
http://support.avaya.com
Toll free number: 1-800-242-2121

For technical support on TELUS system, please contact TELUS technical support at:
http://www.TELUS.com
Toll free number: 1-800-306-1586

3. Reference Configuration

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

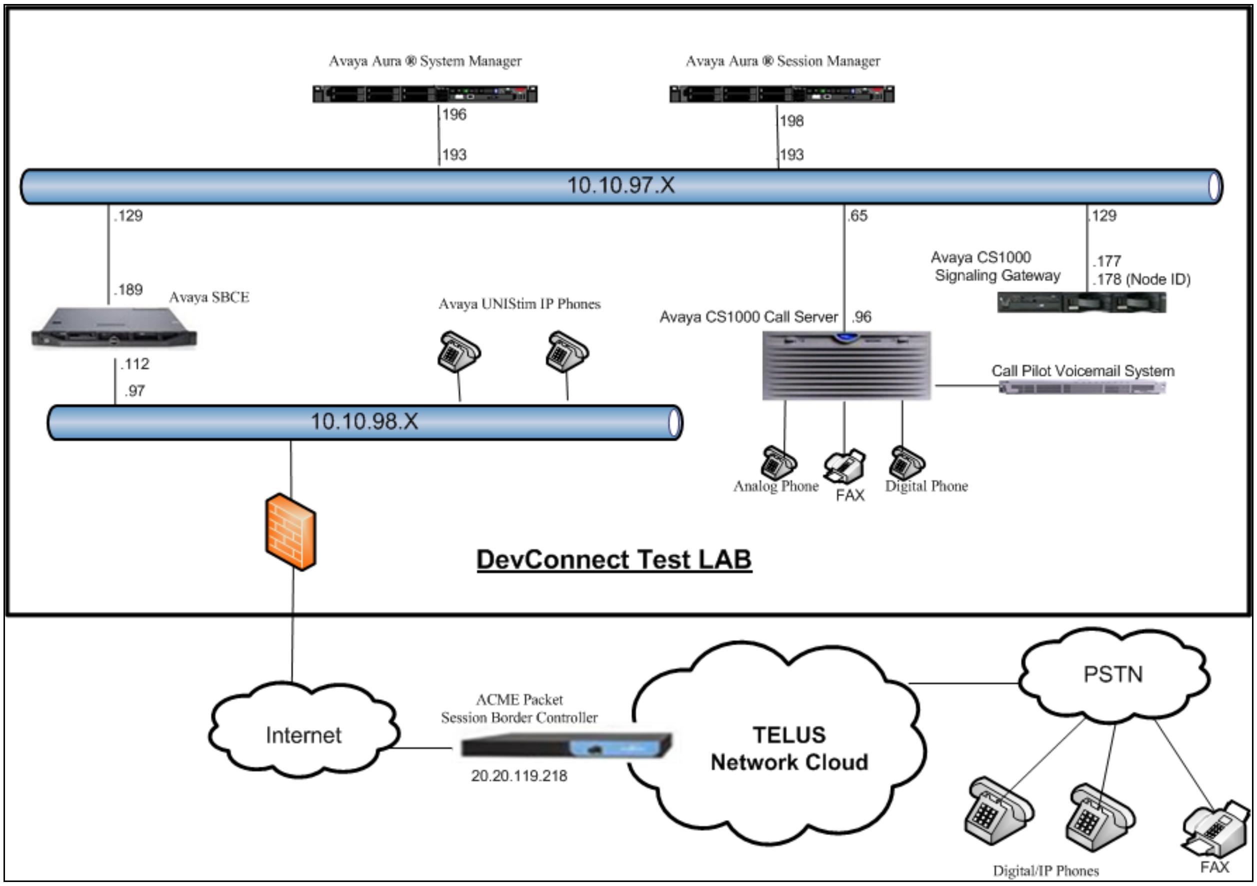


Figure 1- Network diagram for Avaya and TELUS Systems

4. Equipment and Software Validated

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

Avaya system:

System	Software
Avaya Communication Server 1000 (CPPM)	Call Server: 750 Q+ GA Signaling Server: 7.50.17 GA SIP Line Server: 7.50.17 GA
Avaya S8800 Server	Avaya Aura® Session Manager R6.2.0.0.620103 – 6.2.1.621002
Avaya S8800 Server	Avaya Aura ®System Manager R6.2.0 – SP1 – 6.2.0.0.15669 – 6.2.12.105
Avaya Session Border Controller for Enterprise	4.0.5 Q09
Avaya UNIStim Phone	2002 p2: 0604DCN 1140: 0625C8D 1120: 0624C8D 2007: 0621C8D
Avaya 3904 Digital Phone	N/A
Analog Phone	N/A
HP Officejet 4500 Fax	N/A

TELUS system:

System	Software
Acme Packet Net-Net 4250 Session Border Controller	6.1m7p5
Nokia Siemens Networks HiQ 4200	Version 14.0

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

Call Server: 7.50 Q+ GA plus latest DEPLIST – Deplists_CPL_X21_07_50Q.zip
SSG Server: 7.50.17 GA plus latest DEPLIST – Service_Pack_Linux_7.50_17_20120713.ntl
Avaya SBCE: 4.0.5 Q09 plus the patch - HistInfo-mvista-load-Q09.rpm

5. Configure Communication Server 1000

These Application Notes used the Incoming Digit Translation feature to receive the calls and used the Numbering Plan Area Code (NPA), Special Number (SPN) features to route calls from the Communication Server 1000, over the TELUS SIP trunk to PSTN.
These application notes assume that the basic configuration has already been administered. For further information on Communications Server 1000, please consult the references in **Section 10.**

The below procedures describe the configuration details of Communication Server 1000 with a SIP trunk to the TELUS system.

5.1. Log in to Communication Server 1000 System

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

Open an instance of a web browser and connect to the UCM GUI at the following address: http://<node IP address> or http://<UCM IP address>. **Log in** using an appropriate **User ID** and **Password.**

Figure 2 – Login Unified Communications Management

The **Avaya Unified Communications Management** screen is displayed. Click on the **Element Name** of the Communication Server 1000 Element as highlighted in red box as shown in **Figure 3**.

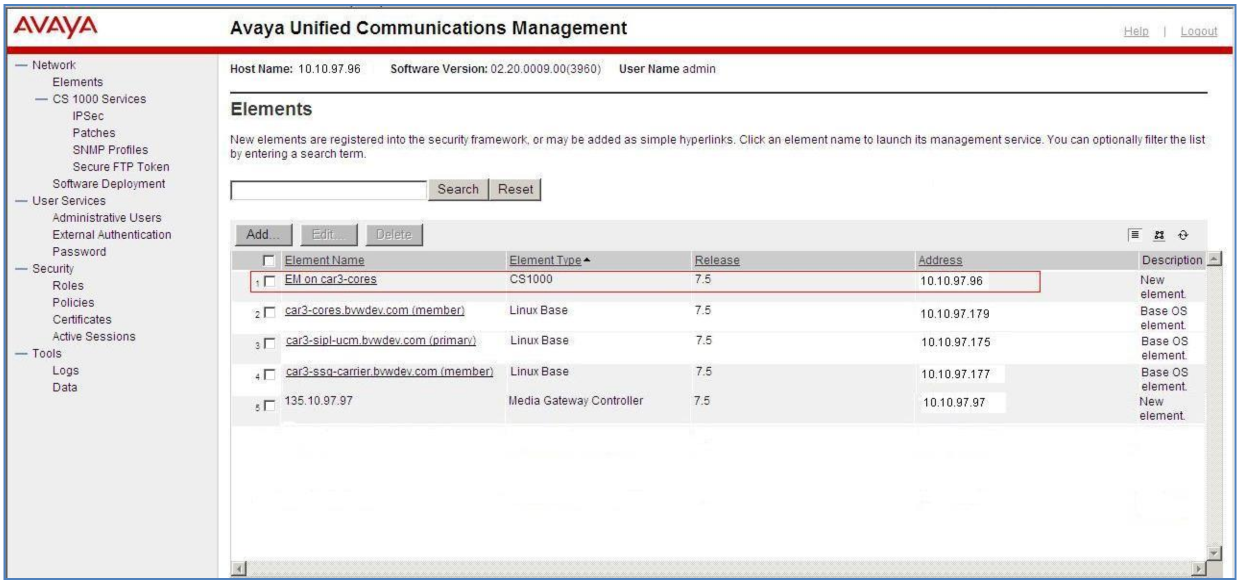


Figure 3 – Unified Communications Management

The Communication Server 1000 Element Manager **System Overview** page is displayed as shown in **Figure 4**.

IP Address: 10.10.97.96
Type: Communication Server 1000E CPPM Linux
Version: 4121
Release: 7.50 Q+

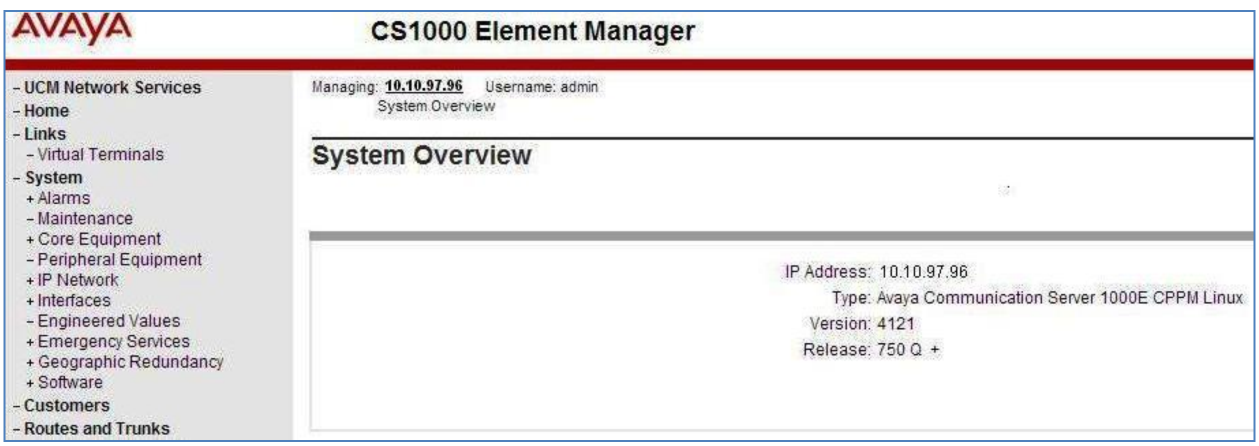


Figure 4 – Element Manager System Overview

5.1.2. Log in to Call Server by using the Overlay Command Line Interface (CLI)

Using Putty, SSH to connect to IP address of SSG Server with the **admin** account.
Run the command **cslogin** and log in with the appropriate **admin** account and password.
Here are the logs.

```
login as: admin

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

admin@10.10.97.177's password: <----enter your password
Last login: Mon July 02 11:42:05 2012 from 10.10.98.78
[admin@car3-ssg-carrier ~]$ cslogin

SEC054 A device has connected to, or disconnected from, a pseudo tty without authenticating
>login
USERID? admin
PASS?  <----enter your password
.
TTY #08 LOGGED IN
The software and data stored on this system are the property of, or licensed to, Nortel Networks
and are lawfully available only to authorized users for approved purposes. Unauthorized access
to any software or data on this system is strictly prohibited and punishable under appropriate
laws. If you are not an authorized user then log out immediately. This system may be monitored
for operational purposes at any time.
 ADMIN 11:43 07/02/2012
>
```

5.2. Administer a Node IP Telephony

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

5.2.1. Obtain Node IP address

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

Select **System → IP Network → Nodes: Servers, Media Cards** and then click on the Node ID as shown in **Figure 5**.

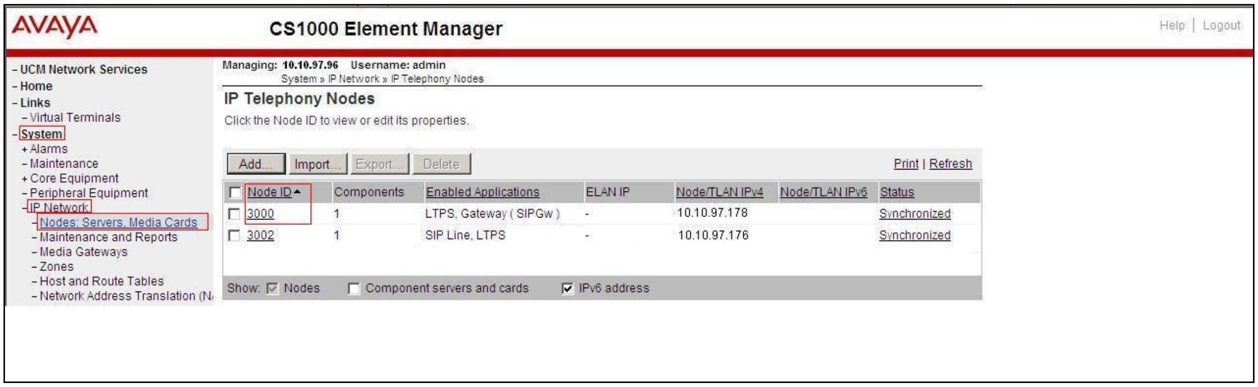


Figure 5 – IP Telephony Nodes

The **Node Details** screen is displayed in **Figure 6 and Figure 7** with the IP address of the Communication Server 1000 node. The **Node IPv4 Address 10.10.97.178** is a virtual address which corresponds to the TLAN IP address **10.10.97.177** of the Signaling Server, SIP Signaling Gateway. The SIP Signaling Gateway uses this Node IP Address to communicate with other components to process SIP calls.

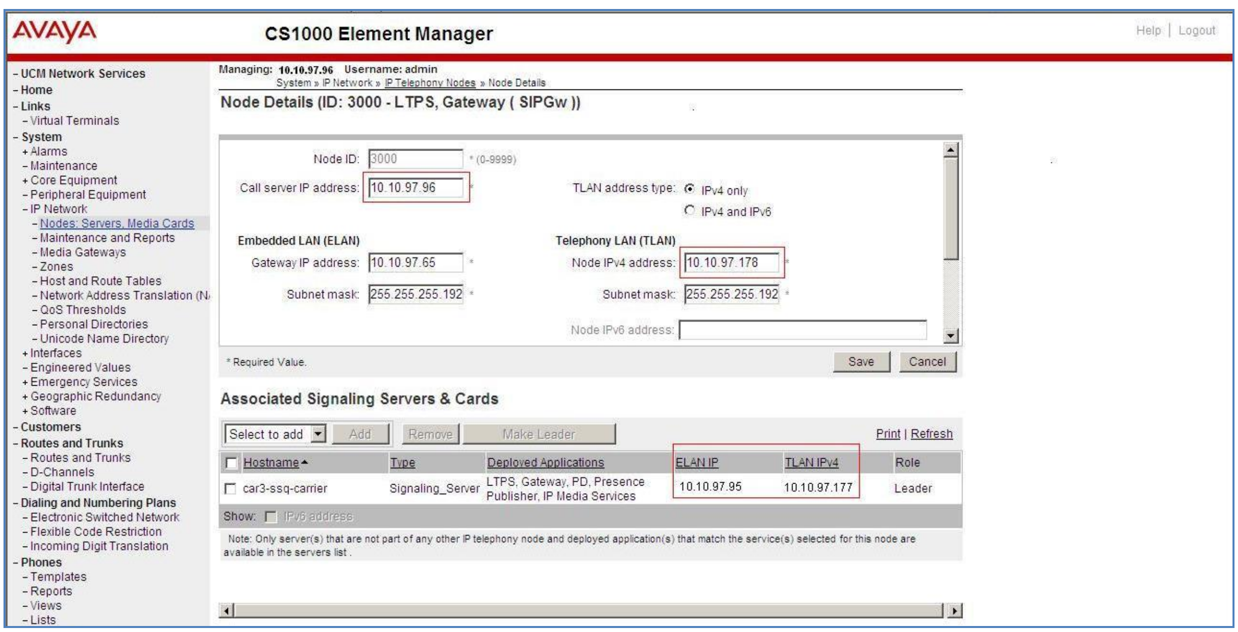


Figure 6 –Node Details

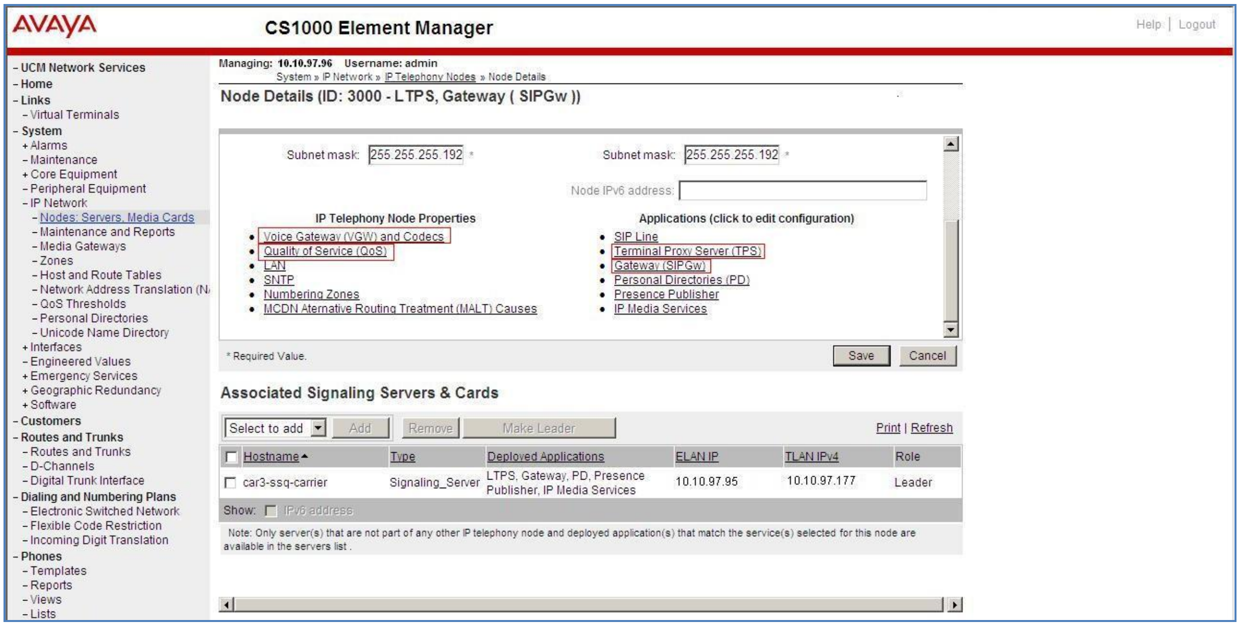


Figure 7 –Node Details

5.2.2. Administer Terminal Proxy Server (TPS)

Continue from **Section 5.2.1**. On the **Node Details** page, select the **Terminal Proxy Server** (**TPS**) link as shown in **Figure 7**.
Check the **UNIStim Line Terminal Proxy Server** checkbox to enable proxy service on this node and then click the **Save** button as shown in **Figure 8**.

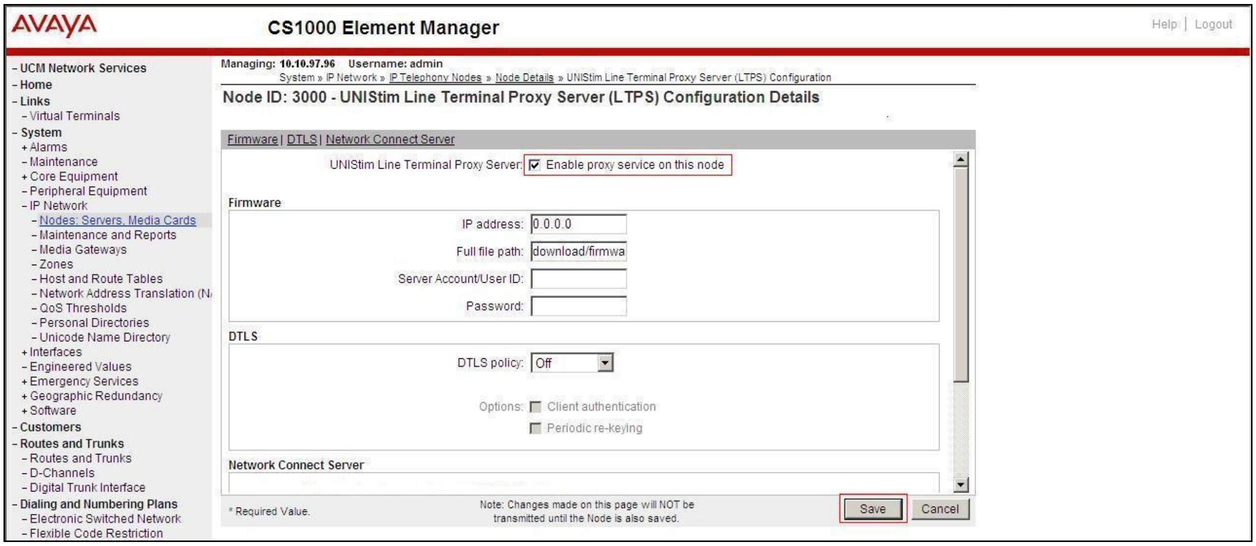


Figure 8 – TPS Configuration Details

5.2.3. Administer Quality of Service (QoS)

Continue from **Section 5.2.1**. On the **Node Details** page, select the **Quality of Service (QoS)** link as shown in **Figure 7**.
The default Diffserv values are as shown in **Figure 9**. Click on the **Save** button.

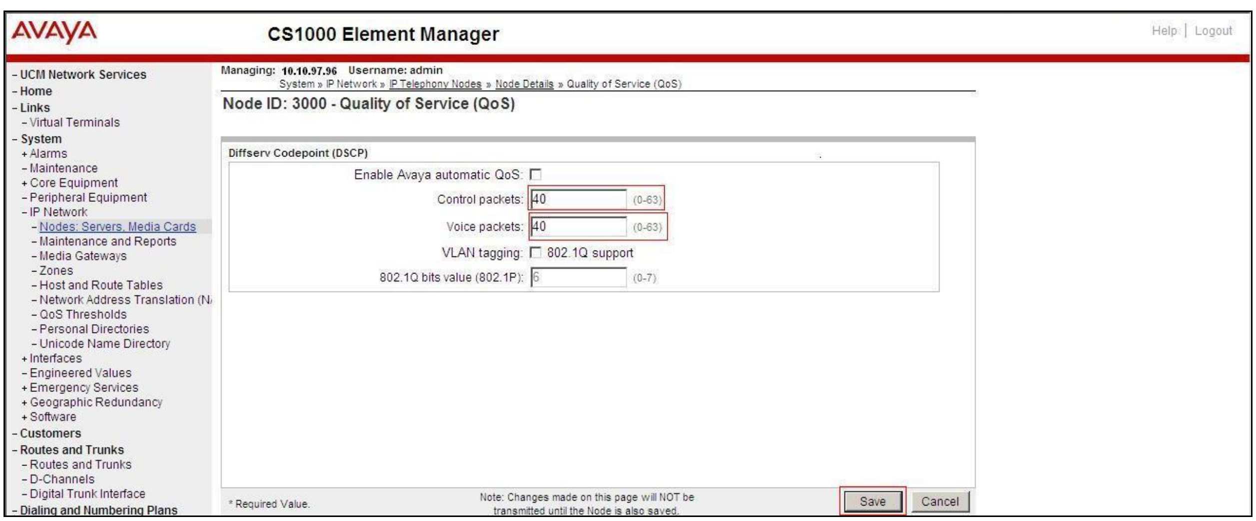


Figure 9 – QoS Configuration Details

5.2.4. Synchronize New Configuration

Continue from **Section 5.2.3**, return to the **Node Details** page (**Figure 6**) and click on the **Save** button.

The **Node Saved** screen is displayed. Click on **Transfer Now** (not shown).

The **Synchronize Configuration Files** screen is displayed. Check the **Signaling Server** checkbox and click on **Start Sync** (not shown).

When the synchronization completes, check the **Signaling Server** checkbox and click on the **Restart Applications** (not shown).

5.3. Administer Voice Codec

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

Select **IP Network → Nodes: Servers, Media Cards** from the left pane, and in the **IP Telephony Nodes** screen displayed, select the **Node ID** of the Communication Server 1000 system. The **Node Details** screen is displayed. (See **Section 5.2.1** for more detail).

On the **Node Details** page as shown in **Figure 7**, click on **Voice Gateway (VGW) and Codecs**. The TELUS system supports **G.711/time 20ms** and **G.729/time 20ms** with **Voice Activity Detection (VAD)** checkbox unchecked. Then click on the **Save** button.

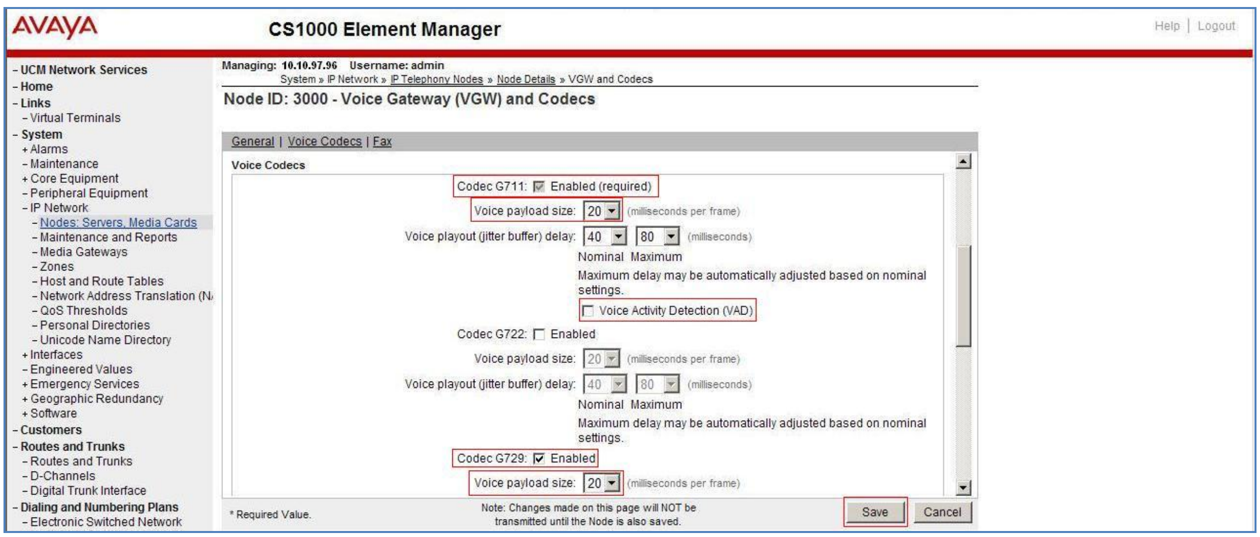


Figure 10 – Voice Gateway and Codec Configuration Details

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

5.3.2. Enable Voice Codec on Media Gateways

From the left menu of the Element Manager page in **Figure 10**, select **IP Network → Media Gateways**. The Media Gateways page will appear (not shown). Click on the **MGC** which is located on the right of the page.

In the following screen scroll down to the Codec **G.711** and **Codec G.729** and uncheck **VAD** as shown in **Figure 11**.

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

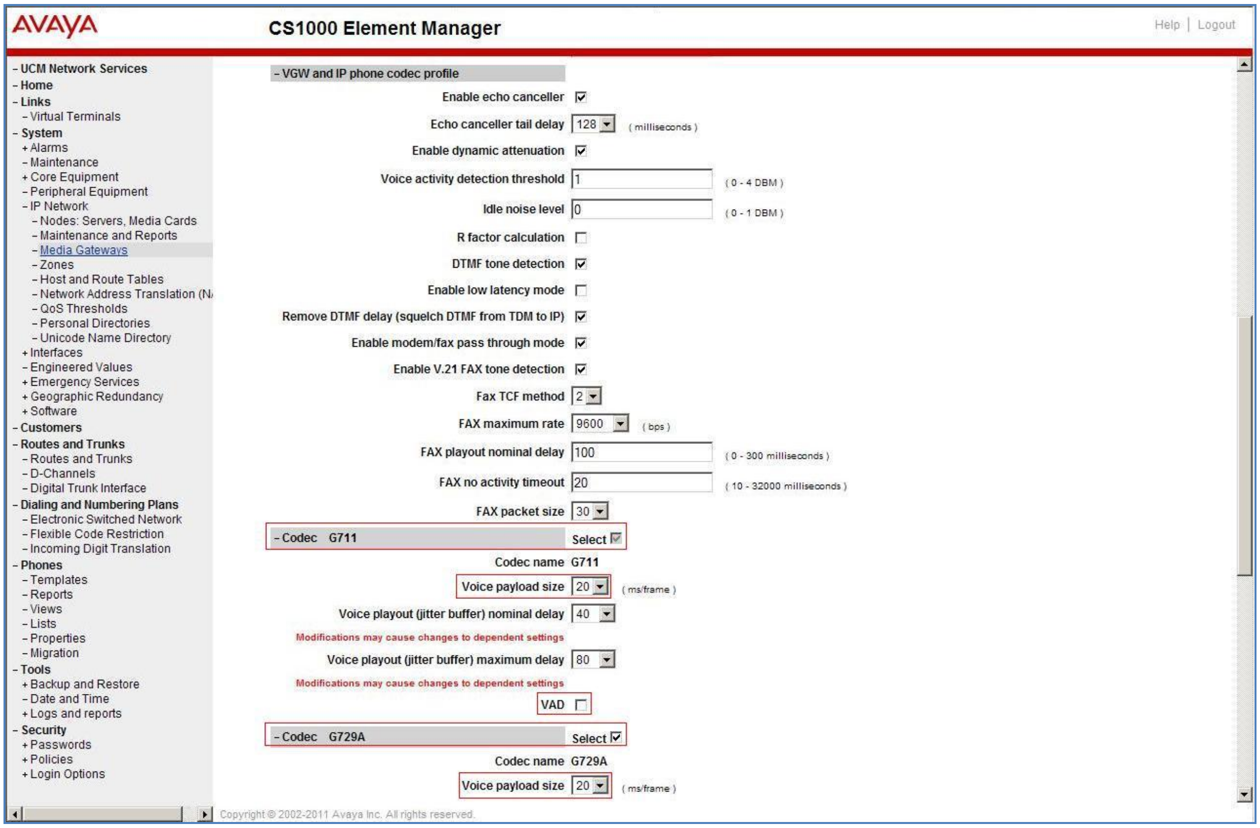


Figure 11 – Media Gateways Configuration Details

5.4. Zones and Bandwidth Management

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

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

The following figures show how to configure a zone for VGW and IP set for bandwidth management purposes. The bandwidth strategy can be adjusted to preference.
Select **IP Network → Zones** configuration from the left pane (not shown), click on **Bandwidth Zones** as shown in **Figure 12**.

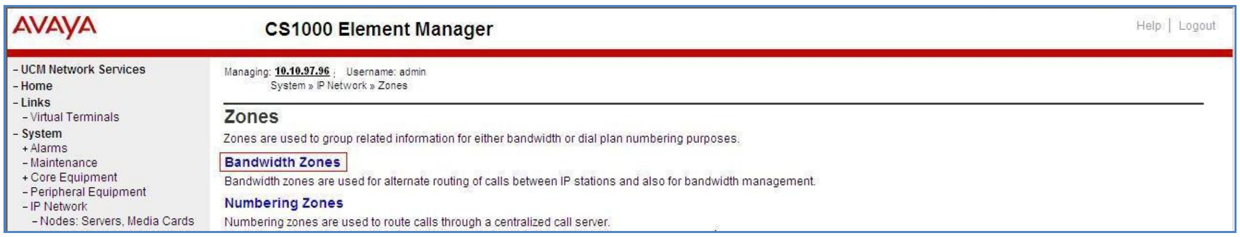


Figure 12 – Zones Page

The **Bandwidth Zones** screen is displayed as shown in **Figure 13**. Click **Add** to create new zone for IP Phones.

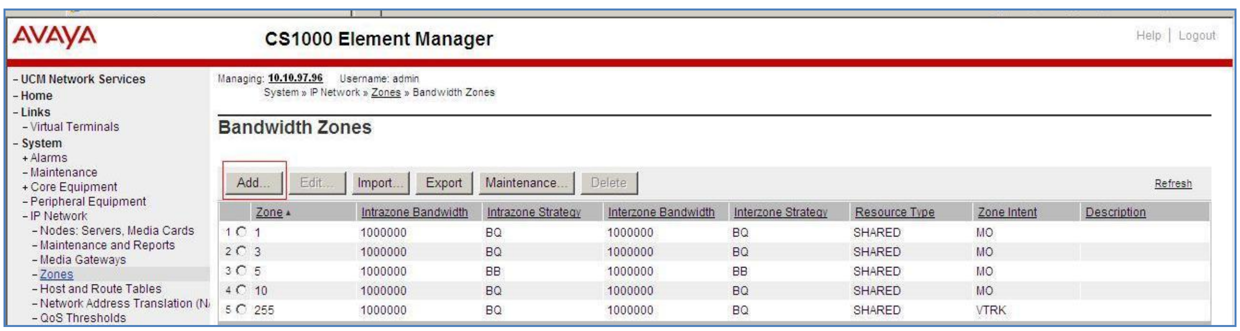


Figure 13 – Bandwidth Zones

Select and input the values as shown (in red box) in **Figure 14** and click on the **Submit** button.

- **INTRA_BW: 1000000**
- **INTRA_STGY**: Set codec for local calls. Select **Best Quality (BQ)** to use G.711 as the first priority codec negotiation.
- **INTER_BW: 1000000** INTER_STGY: Set codec for the calls over trunk. Select **Best Quality (BQ)** to use G.711 as the first priority codec negotiation.
- **Zone Intent** (**(ZBRN)**): Select **MO (MO)** for IP phones, VGW

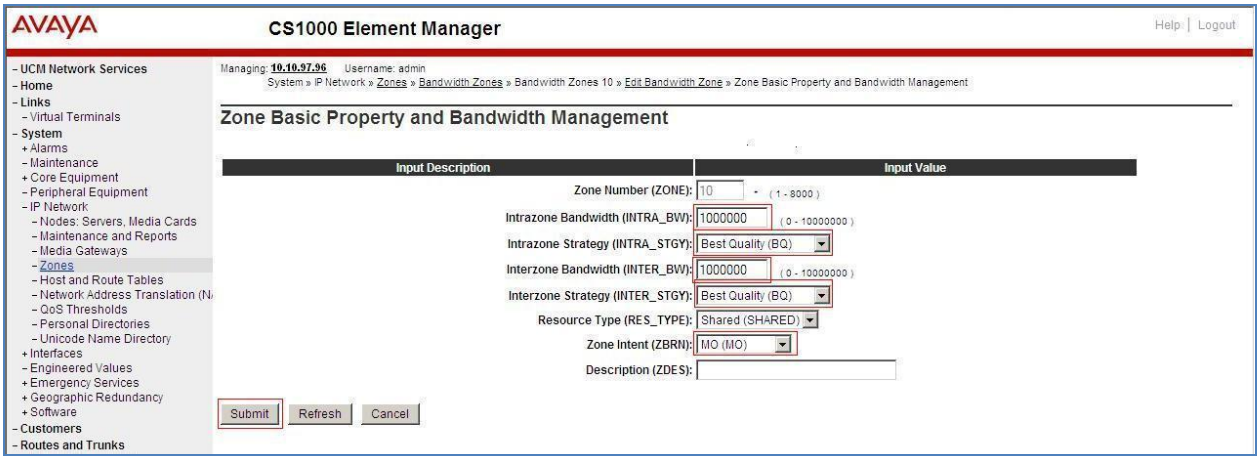


Figure 14 –Bandwidth Management Configuration Details – IP phone

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

Follow **Section 5.4.1** to create a zone for the virtual trunk. The difference is in **Zone Intent (ZBRN)** field. Select **VTRK** for virtual trunk as shown in **Figure 15** and then click on the **Submit** button.

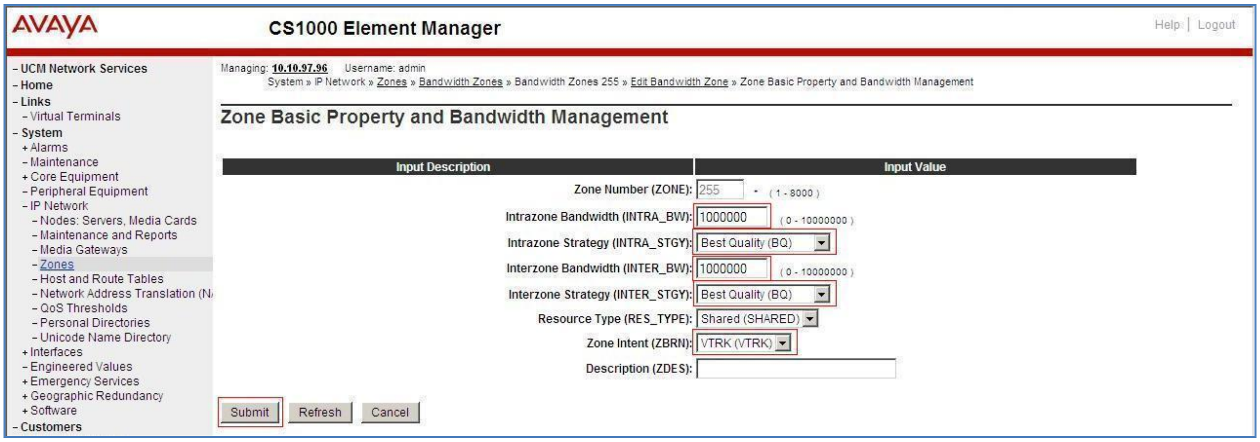


Figure 15 –Bandwidth Management Configuration Details –virtual SIP trunk

5.5. Administer SIP Trunk Gateway

This section describes the steps for establishing a SIP connection between SIP Signaling Gateway (SSG) to Avaya Aura® Session Manager.

5.5.1. Integrated Services Digital Network (ISDN)

Select **Customers** in the left pane (not shown). The **Customers** screen is displayed. Click on the link associated with the appropriate customer, in this case **00**. The system can support more than one customer with different network settings and options. The **Customer 00 Edit** page will appear (not shown). Select the **Feature Packages** option from **Customer 00 Edit** page.
The screen is updated with a listing of feature packages populated below **Feature Packages** (not all features shown in **Figure 16** below). Select **Integrated Services Digital Network** to edit its parameters. The screen is updated with parameters populated below **Integrated Services Digital Network**. Click on **Integrated Services Digital Network**, and retain the default values for all remaining fields. Scroll down to the bottom of the screen, and click on the **Save** button at the bottom of the page (not shown).

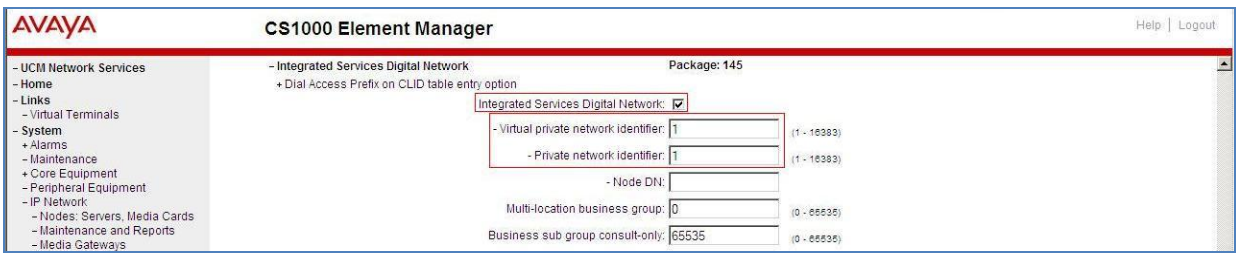


Figure 16 –Customer – ISDN Configuration

5.5.2. Administer SIP Trunk Gateway to Avaya Aura® Session Manager

Select **IP Network → Nodes: Servers, Media Cards** configuration from the left pane. In the **IP Telephony Nodes** screen displayed, select the **Node ID** of the Communication Server 1000 system. The **Node Details** screen is displayed as shown in **Figure 7**, **Section 5.2.1.**

On the **Node Details** screen, select **Gateway (SIPGw)**.

Under the **General** tab of the **Virtual Trunk Gateway Configuration Details** screen, enter the following values (highlighted in red boxes) for the specified fields, and retain the default values for the remaining fields as shown in **Figure 17**. The parameters (highlighted in red boxes) are filled in. The **SIP domain name** and **Local SIP port** should be matched in Avaya Aura® Session Manager configuration **(**in **Section 6.1)**.

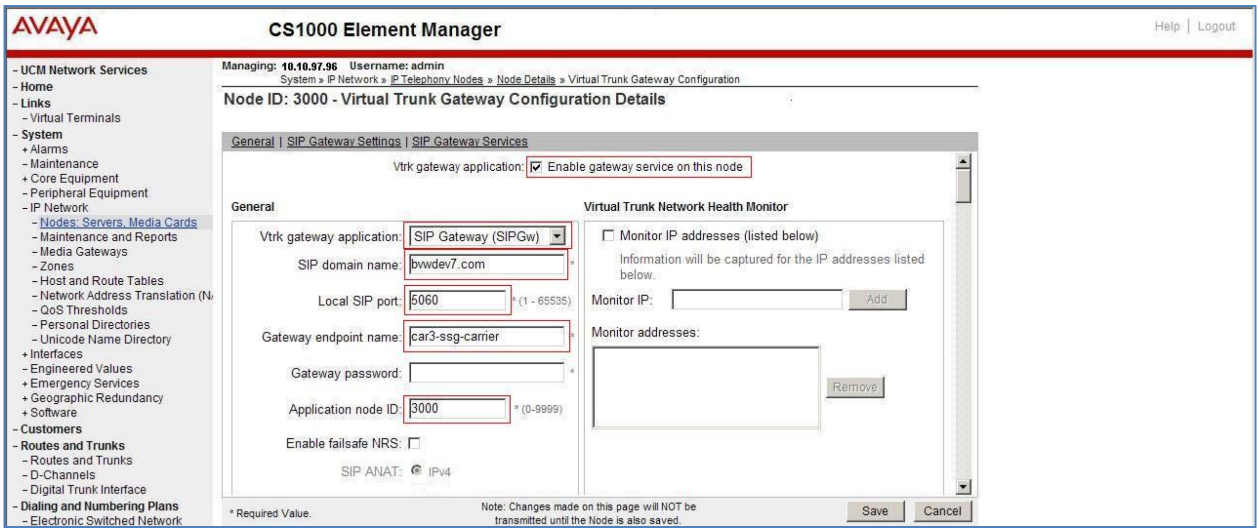


Figure 17 – Virtual Trunk Gateway Configuration Details

Click on the **SIP Gateway Settings** tab, under **Proxy or Redirect Server**, enter the following values (highlighted in red boxes) for the specified fields, and retain the default values for the remaining fields as shown in **Figure 18**. Enter **Primary TLAN IP address** as the IP address of Avaya Aura® Session Manager. Enter Port: **5060** and Transport protocol: **UDP**. Uncheck **Support registration** checkbox.

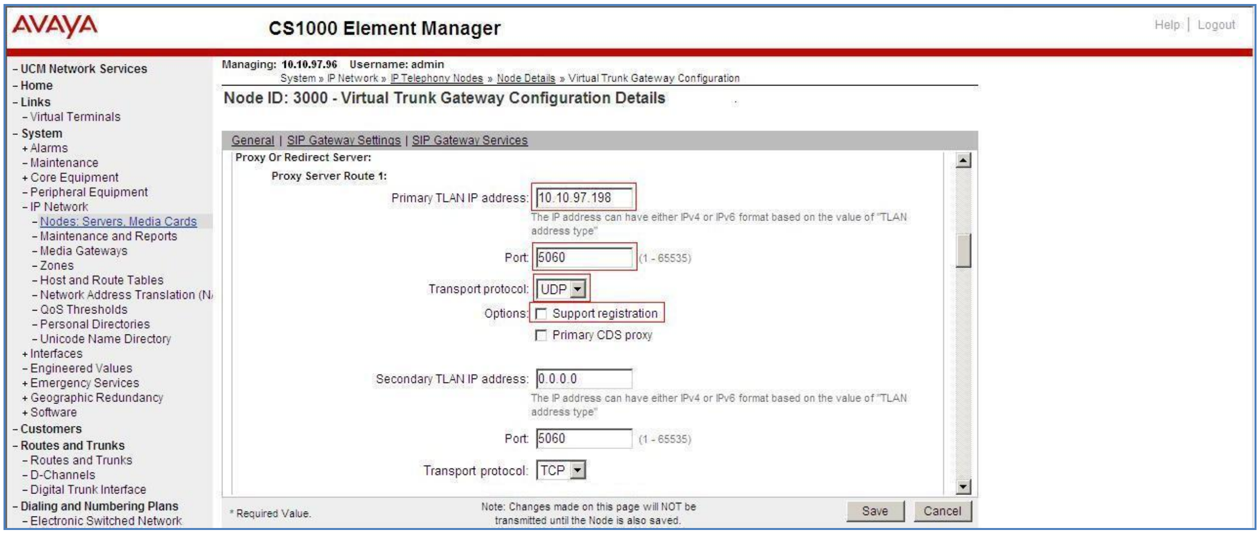


Figure 18 – Virtual Trunk Gateway Configuration Details

On the same page as shown in **Figure 18**, scroll down to the **SIP URI Map** section.
Under the **Public E.164 Domain Names**, for:

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

Under the **Private domain names**, for:

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

The remaining fields can be left at their default values as shown in **Figure 19**. Then click on the **Save** button.

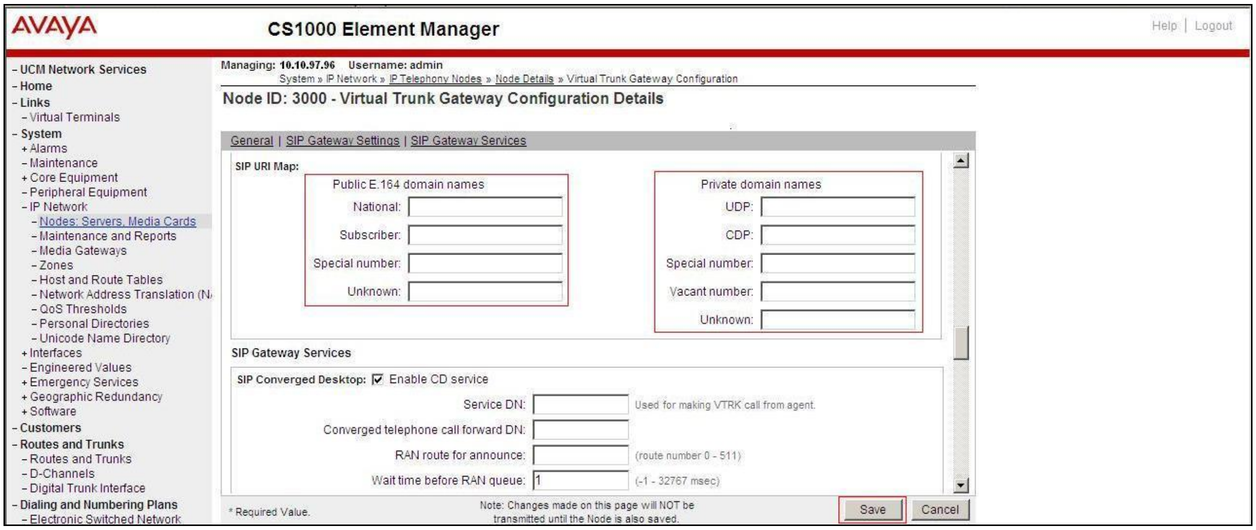


Figure 19 – Virtual Trunk Gateway Configuration Details

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

5.5.3. Administer Virtual D-Channel

Select **Routes and Trunks → D-Channels** (not shown) from the left pane to display the **D-Channels** screen. In the **Choose a D-Channel Number** field, select an available D-channel from the drop-down list and type DCH as shown in **Figure 20**. Click the **to Add** button.

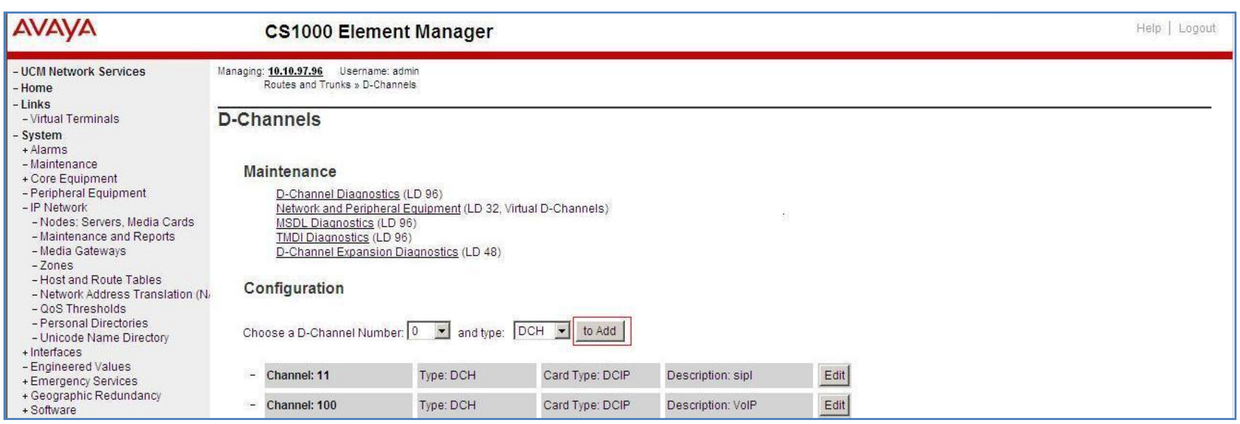


Figure 20 – D-Channels

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

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

Click on **Advanced options (ADVOPT)**. Check on the **Network Attendant Service Allowed** checkbox as shown in **Figure 21**. Other fields are left as default.

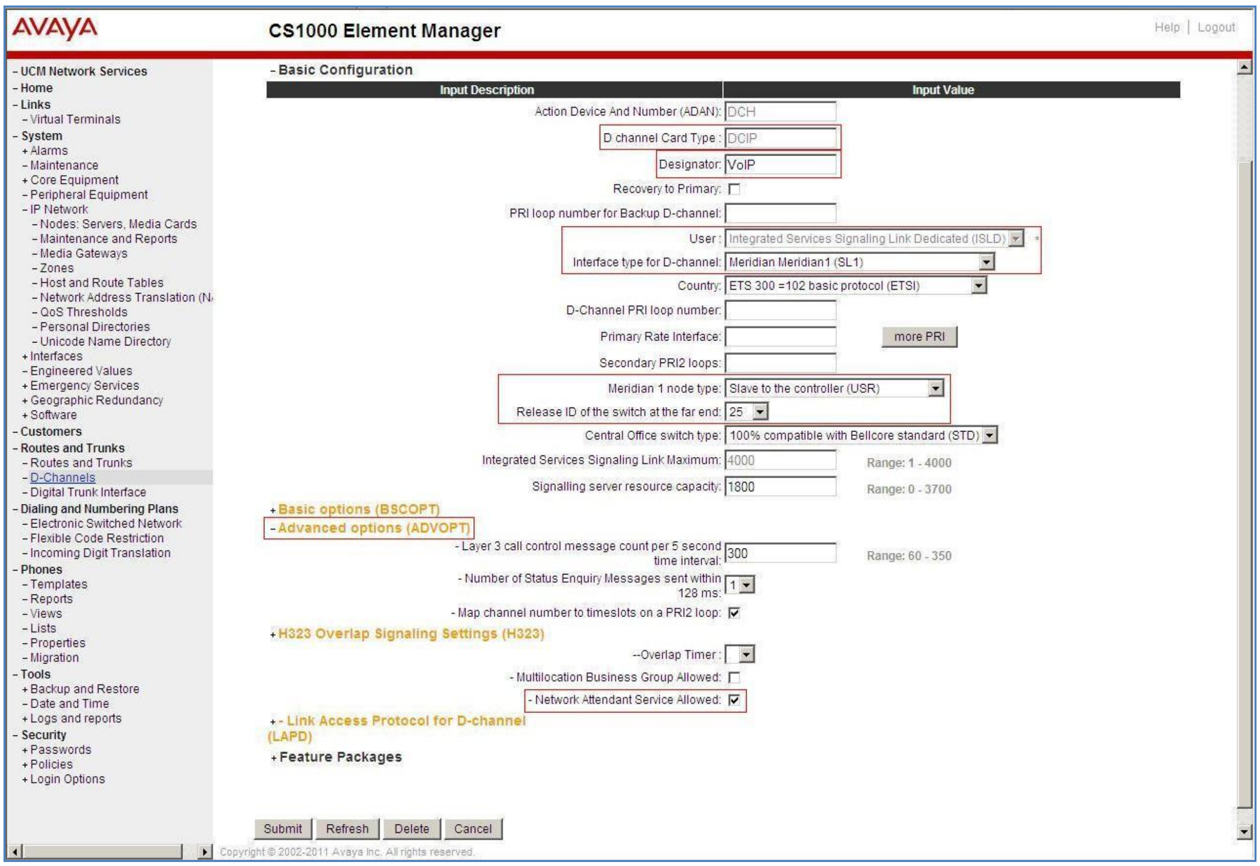


Figure 21 – D-Channels Configuration Details

Click on the **Basic Options (BSCOPT)** and click on the **Edit** button on the **Remote Capabilities** field. The **Remote Capabilities Configuration** page will appear. Check on the **ND2** and the **MWI** checkboxes as shown in **Figures 23**.

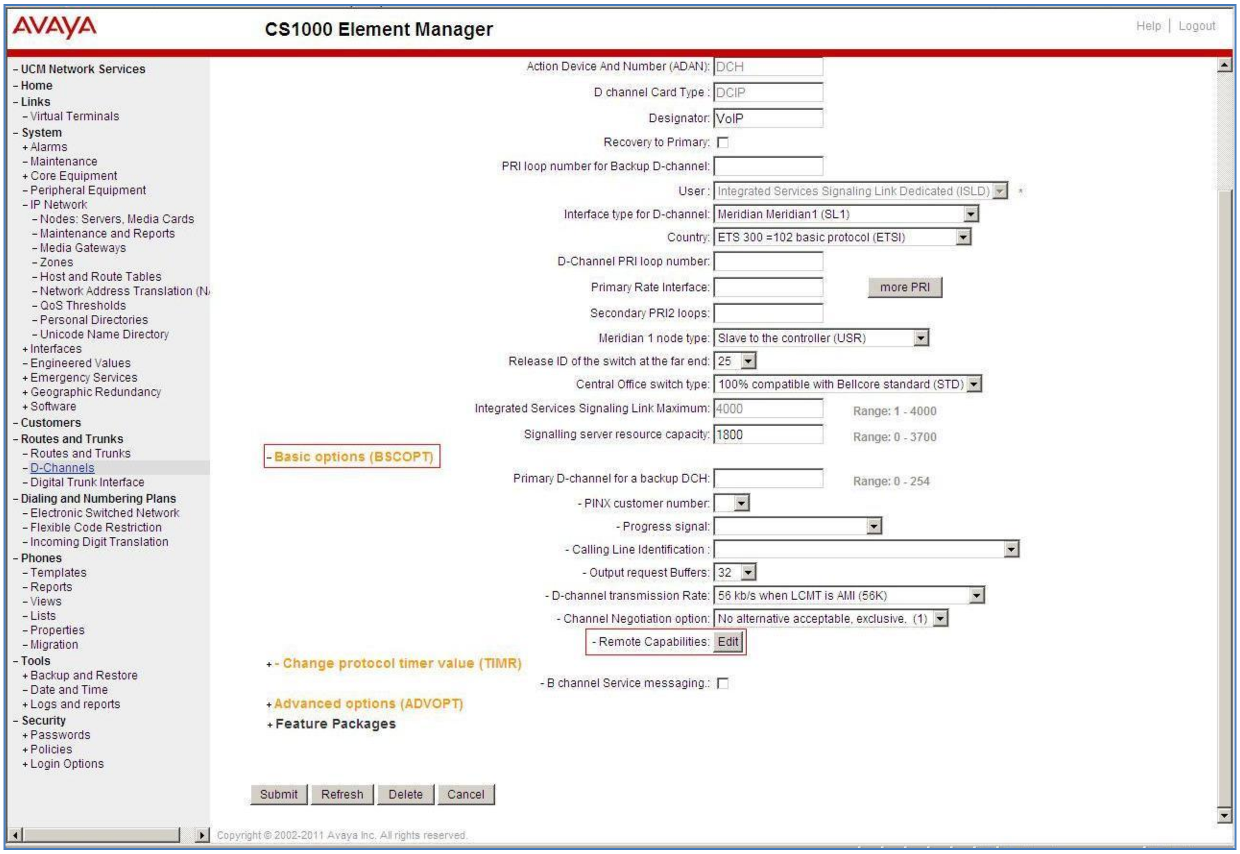


Figure 22 – D-Channel Configuration Details

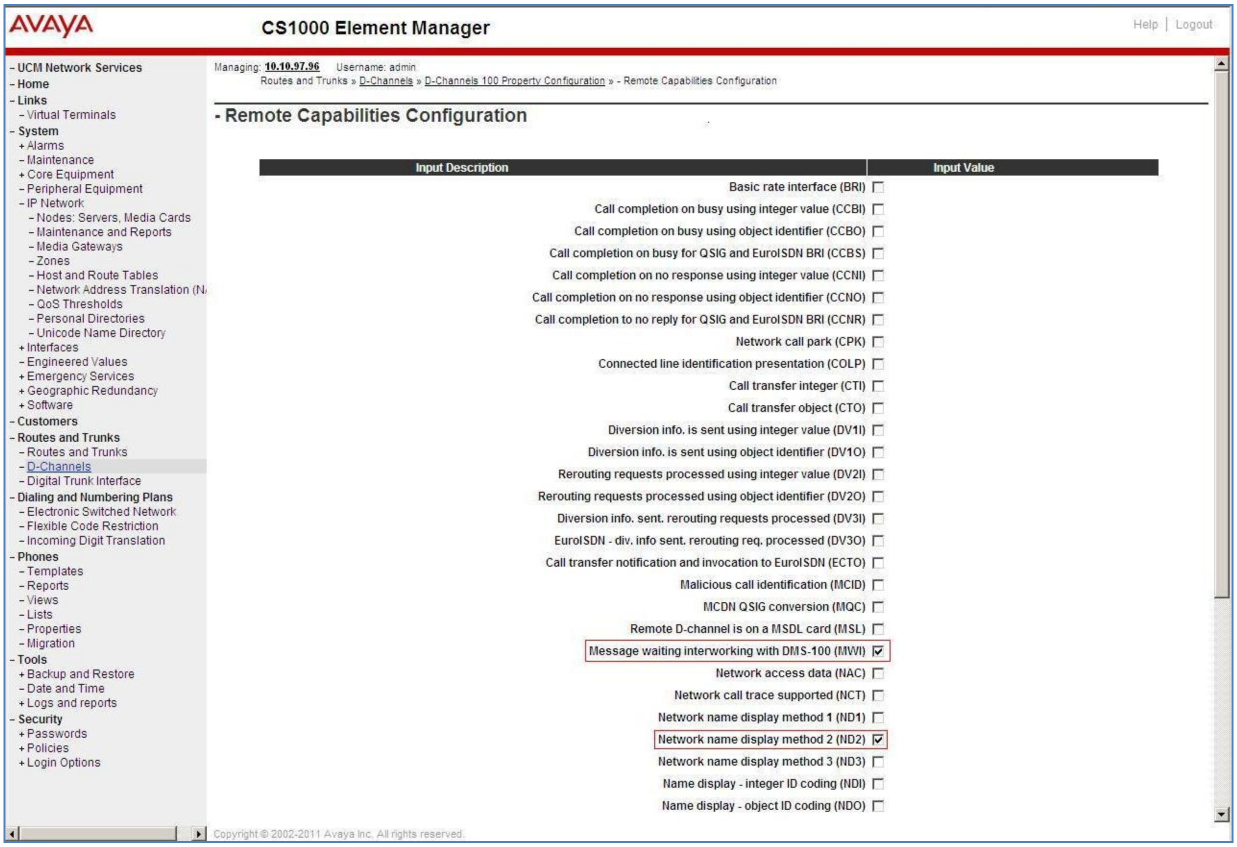


Figure 23 – Remote Capabilities Configuration Details

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

5.5.4. Administer Virtual Super-Loop

Select **System → Core Equipment → Superloops** from the left pane to display the **Superloops** screen. If the Superloop does not exist, please click the **Add** button to create a new one as shown in **Figure 24**. In this example, Superloop 4, 96, 100 and 124 have been added and are being used.

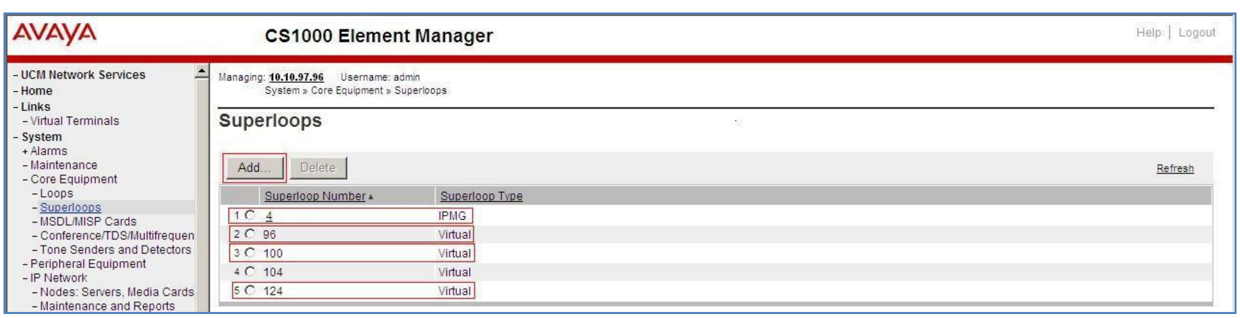


Figure 24 – Administer Virtual Super-Loop Page

5.5.5. Administer Virtual SIP Routes

Select **Routes and Trunks → Routes and Trunks** (not shown) from the left pane to display the **Routes and Trunks** screen. In this example, **Customer 0** is being used. Click on the **Add route** button as shown in **Figure 25**.

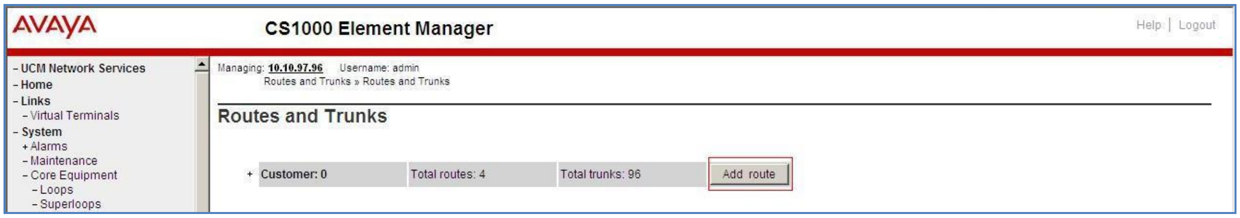


Figure 25 – Add route

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

- **Route number** (ROUT): Select an available route number (example: route 100).
- **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 **The route is for a virtual trunk route** (VTRK) field, to enable four additional fields to appear.
- For the **Zone for codec selection and bandwidth management** (ZONE) field, enter 255 (created in **Section 5.4.2**).
- For the **Node ID of signaling server of this route** (NODE) field, enter the node number 3000 (created in **Section 5.2.1**).
- Select **SIP (SIP)** from the drop-down list for the **Protocol ID for the route (PCID)** field.
- Check the **Integrated Services Digital Network option** (ISDN) checkbox to enable additional fields to appear. Enter the following values for the specified fields, and retain the default values for the remaining fields. Scroll down to the bottom of the screen.
  - **Mode of operation** (MODE): Select Route uses ISDN Signalling Link (ISLD)
  - **D channel number** (DCH): Enter 100 (created in **Section 5.5.3**)
  - **Network calling name allowed** (NCNA): Check the field.
  - **Network call redirection** (NCRD): Check the field.
  - **Insert ESN access code** (INAC)**:** Check the field.

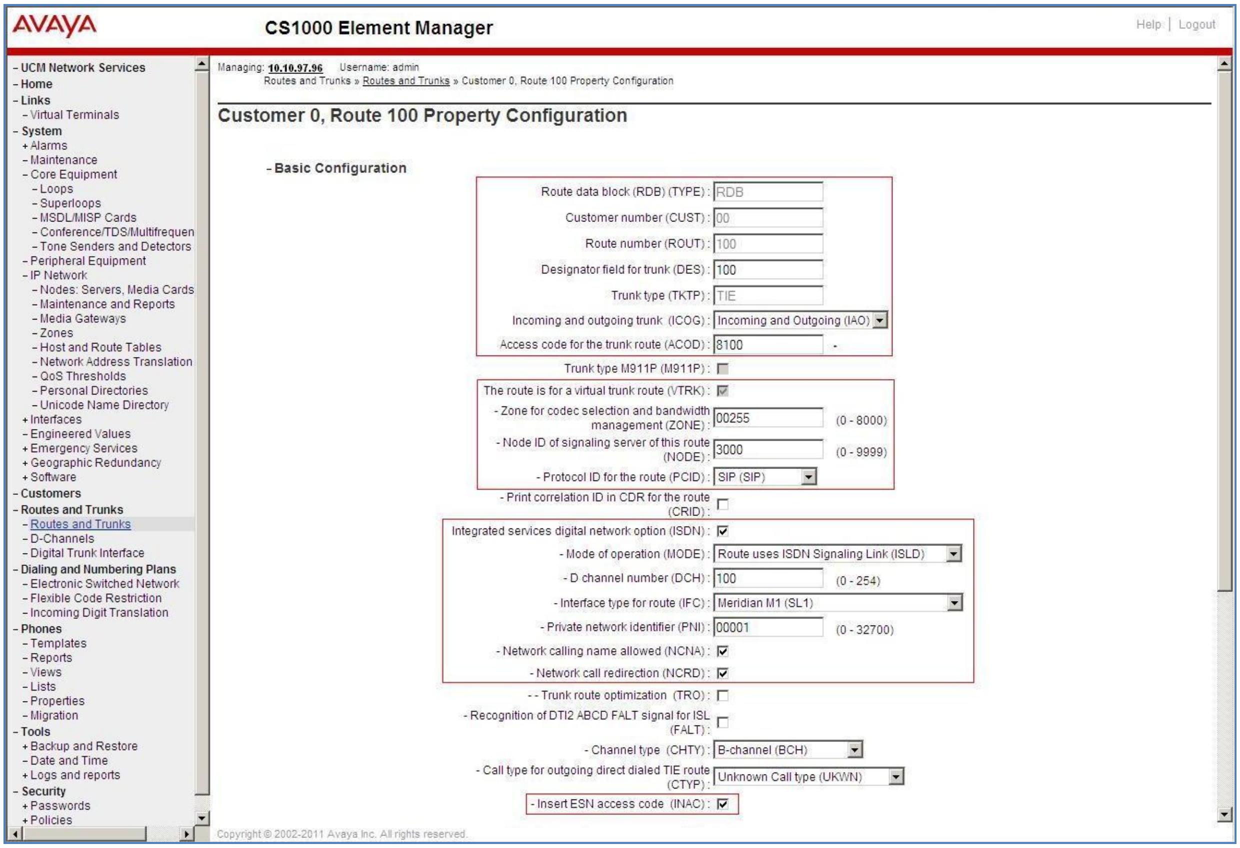


Figure 26 – Route Configuration Details

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

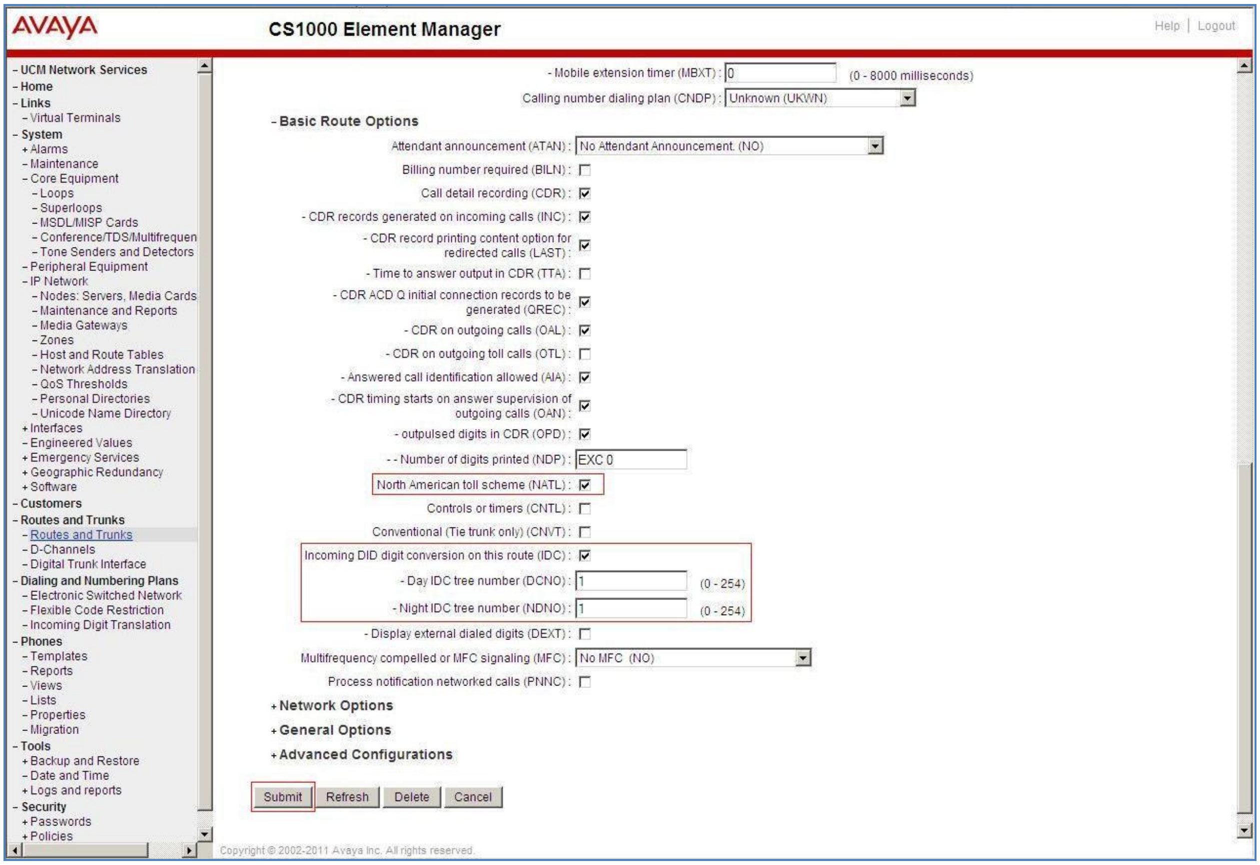


Figure 27 – Route Configuration Details

5.5.6. Administer Virtual Trunks

Select **Routes and Trunks → Route and Trunks** (not shown). The Route list is now updated with the newly added routes. In the example, the Route 100 was being added. Click on the **Add trunk** button as shown in **Figure 28**.

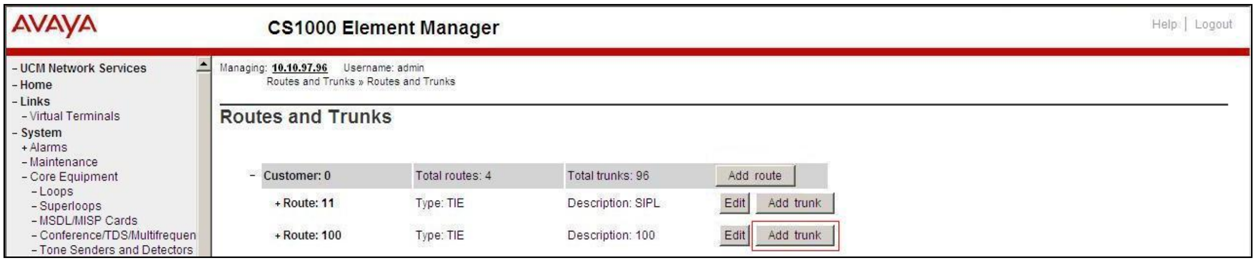


Figure 28 – Routes and Trunks Page

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

- The Multiple trunk input number (MTINPUT) field may be used to add multiple trunks in a single operation, or repeat the operation for each trunk. In the sample configuration, 32 trunks were created.
- Trunk data block (**TYPE**): IP Trunk (IPTI)
- Terminal Number (**TN**): Available terminal number (created in **Section 5.5.4**)
- Designator field for trunk (**DES**): A descriptive text
- Extended Trunk (**XTRK**): Virtual trunk (VTRK)
- Member number (**RTMB**): Current route number and starting member
- Card Density: 8D
- Start arrangement Incoming (**STRI**): Immediate (IMM)
- Start arrangement Outgoing (**STR**O): Immediate (IMM)
- Trunk group access restriction (**TGAR**): Desired trunk group access restriction level
- Channel ID for this trunk (**CHID**): An available starting channel ID

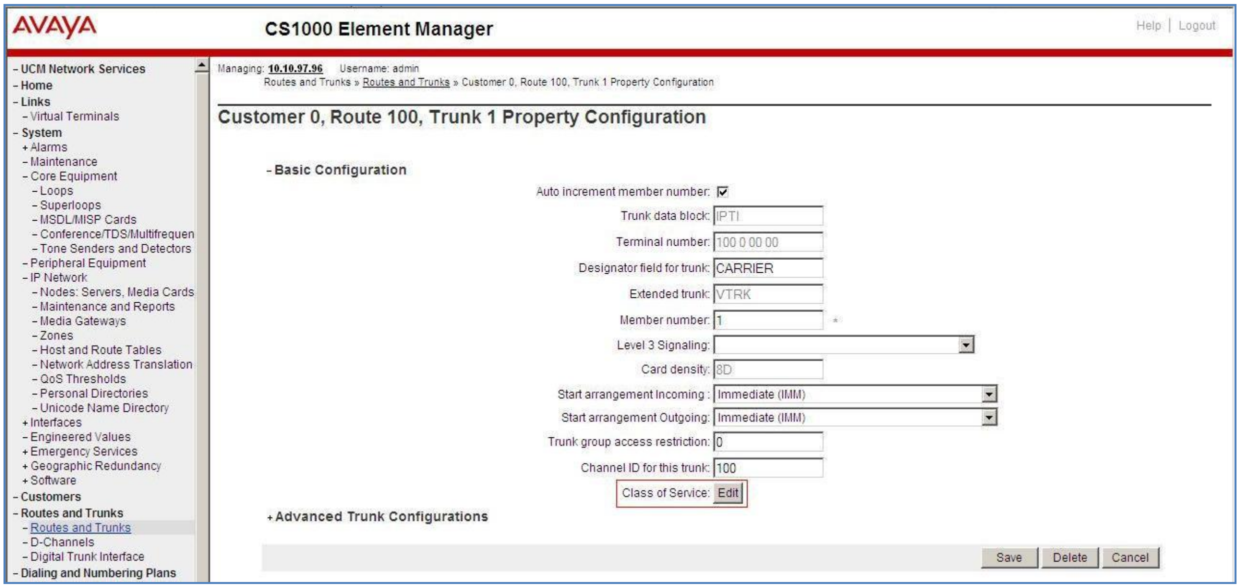


Figure 29 – New Trunk Configuration Details

For **Media Security**, select **Media Security Never** (**MSNV)**. Enter the remaining values for the specified fields as shown in **Figure 30**. Scroll down to the bottom of the screen and click **Return Class of Service** and click on the **Save** button (not shown).

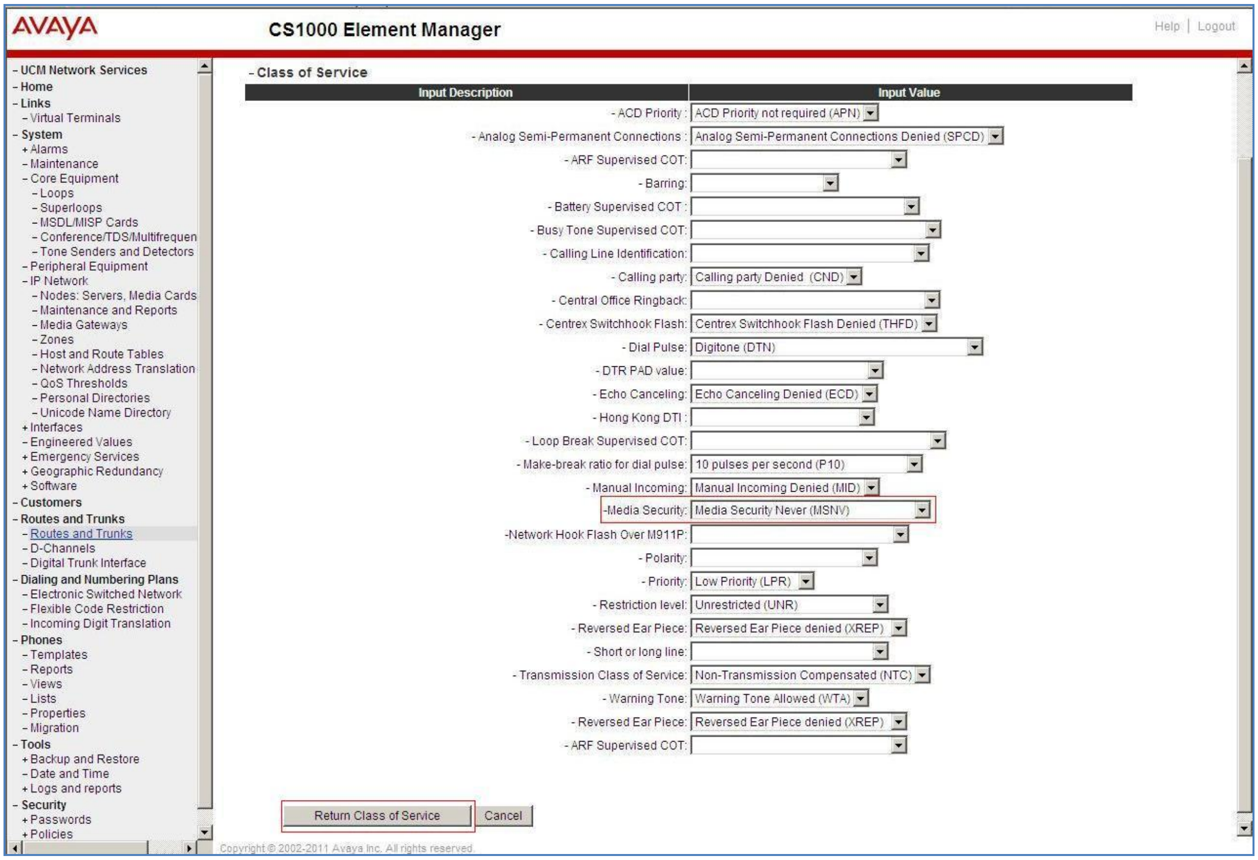


Figure 30 – Class of Service Configuration Details Page

5.5.7. Administer Calling Line Identification Entries

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

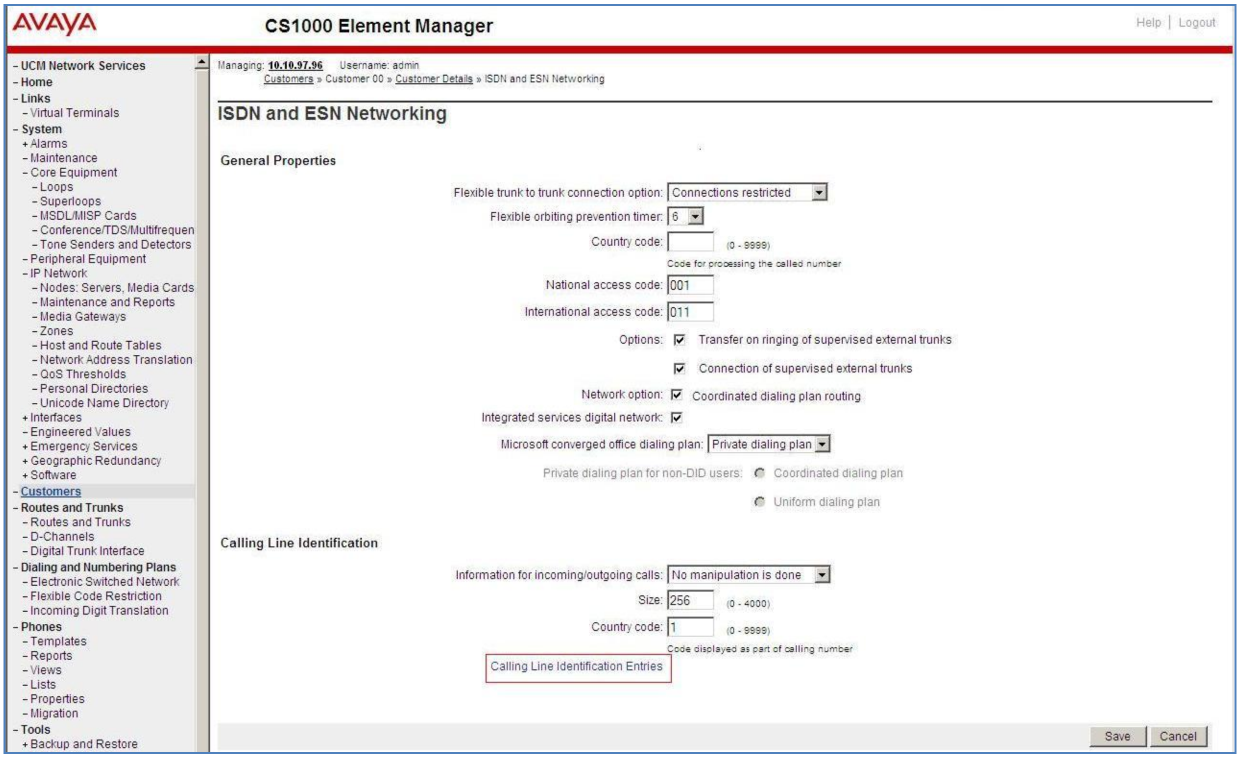


Figure 31 – ISDN and ESN Networking

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

Figure 32 – Calling Line Identification Entries

Add entry **0** as shown in **Figure 33:**

- **National Code**: input prefix digits assigned by Service Provider, in this case it is 3 digits – 403.
- **Local Code**: input prefix digits assigned by Service Provider, in this case it is 3 digits – 692. This **Local Code** will be used for call display purpose for Call Type = Unknown.
- **Home Location Code**: input prefix digits assigned by Service Provider, in this case it is 6 digits - 403692. This **Home Location Code** will be used for call display purpose for Call Type = National (NPA).
- **Local Steering Code**: input prefix digits assigned by Service Provider, in this case it is 6 digits - 403692. This **Local Steering Code** will be used for call display purpose for Call Type = Local Subscriber (NXX).
- **Calling Party Name Display**: Uncheck for **Roman characters**.

Click on the **Save** button as shown in **Figure 33**.

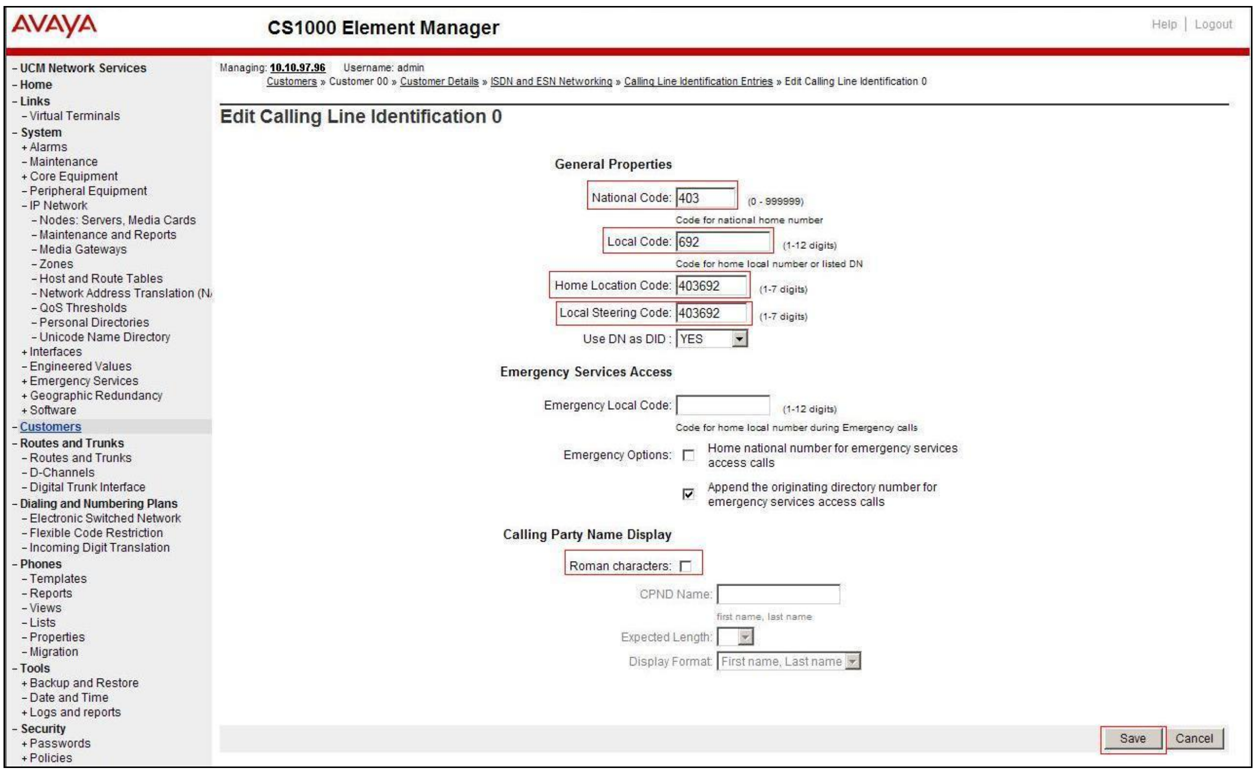


Figure 33 – Edit Calling Line Identification 0

5.5.8. Enable External Trunk to Trunk Transfer

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

- Login Call Server Overlay CLI (please refer to **Section 5.1.2** for more detail).
- Allow External Trunk to Trunk Transfer for Customer Data Block by using **LD 15**.

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

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

5.6. Administer Dialing Plans

5.6.1. Define ESN Access Codes and Parameters (ESN)

Select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (**ESN**) screen as shown in **Figure 34**.

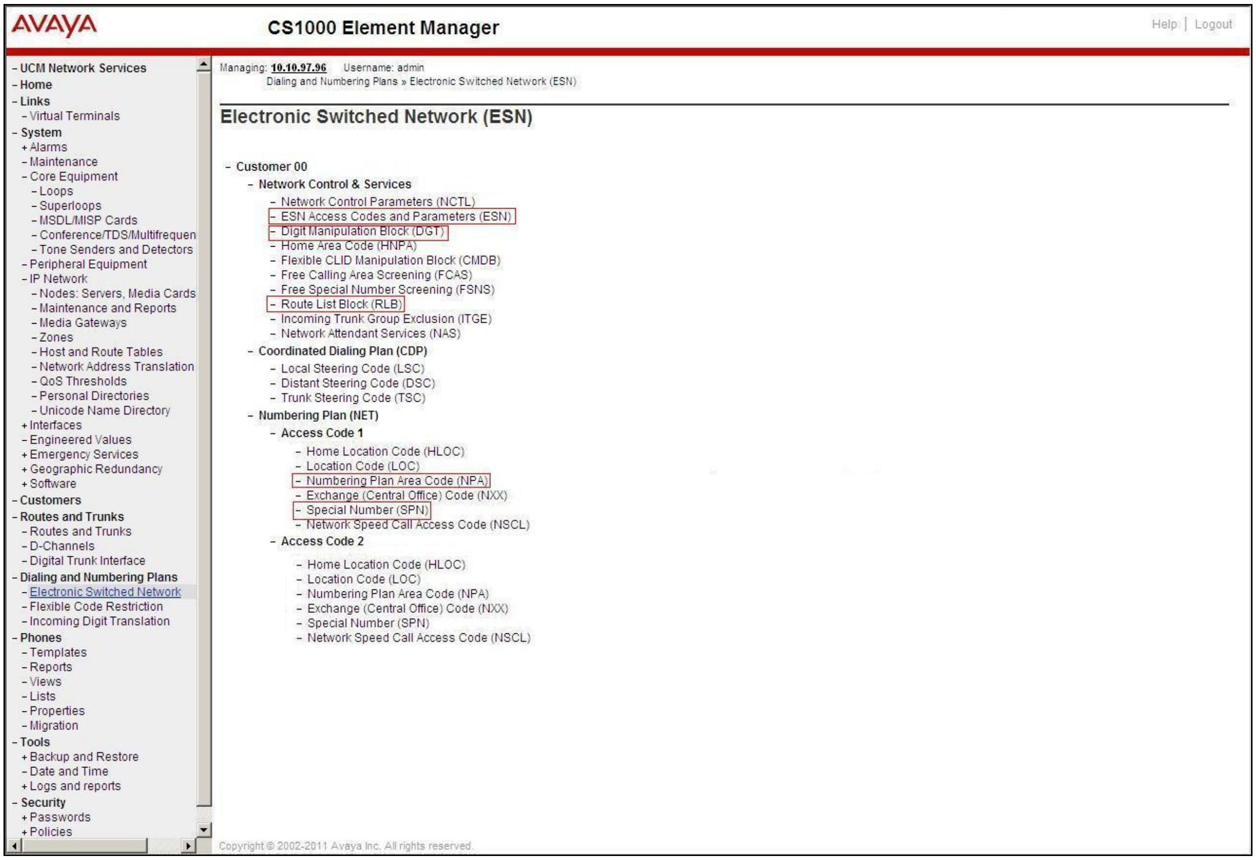


Figure 34 –ESN Configuration Details

In the **ESN Access Codes and Basic Parameters** page, define **NARS/BARS Access Code 1** as shown in **Figure 35**.
Click the **Submit** button (not shown).

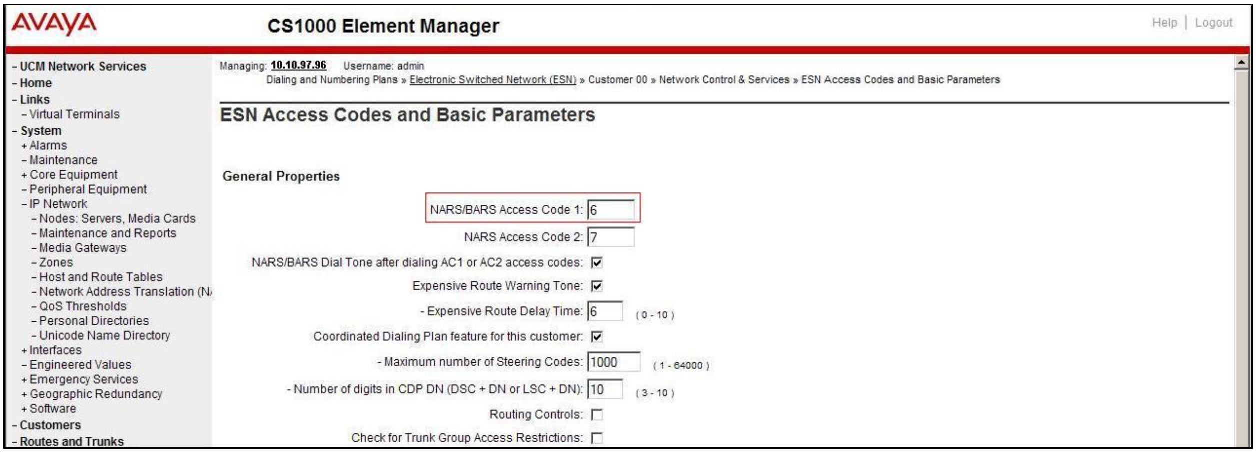


Figure 35 – ESN Access Codes and Basic Parameters

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

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

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

TYPE NET_DATA
CUST 0
OPT
AC2 xNPA xSPN     → (Set NPA, SPN not to associate to ESN Access Code 2)
FNP
CLID
…
```

Verify Customer Net Data block by using LD 21.

```
>ld 21
PT1000

REQ: prt
TYPE: net
TYPE NET_DATA
CUST 0

TYPE NET_DATA
CUST 00
OPT RTA
AC1 INTL NPA SPN NXX LOC ------  > (NPA, SPN are associated to ESN Access Code 1)
AC2
FNP YES
…
```

5.6.3. Digit Manipulation Block (DMI)

Select **Dialing and Numbering Plans → Electronic Switched Network** (not shown) from the left pane to display the **Electronic Switched Network** (ESN) screen. Select **Digit Manipulation Block** (DGT) as shown in **Figure 34**

Select an available DMI from the drop-down list and click **to Add** as shown in **Figure 36**

Enter the **Number of leading digits to be Deleted** (Del) field and select the **Call Type to be used by the manipulated digits** (CTYP) and then click Submit (see **Figure 37, Figure 38**).

5.6.4. Digit Manipulation Block Index (DMI) for Outbound Call

The following steps show how to add DMI for the outbound call. There are 2 indexes, which were added to the Digit Manipulation Block List (14 and 15).

Select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen. Select **Digit Manipulation Block** (DGT).

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

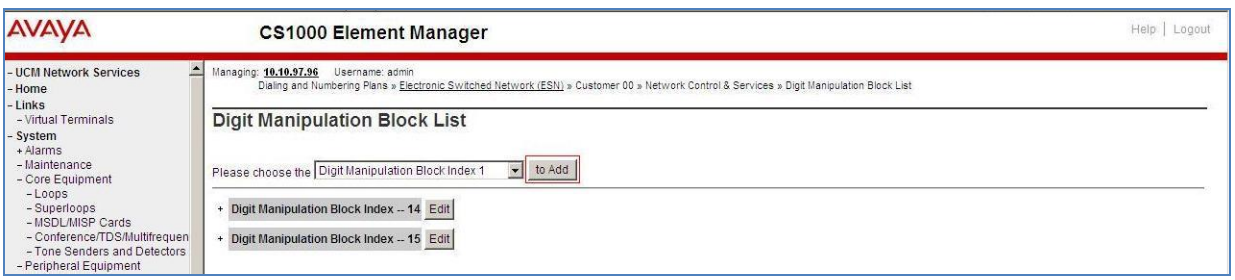


Figure 36 – Add a DMI

Add DMI_14: Enter **0** for the **Number of leading digits to be Deleted** (Del) field and select **NPA** for the **Call Type to be used by the manipulated digits** (CTYP) and then click on **Submit** button as shown in **Figure 37**.

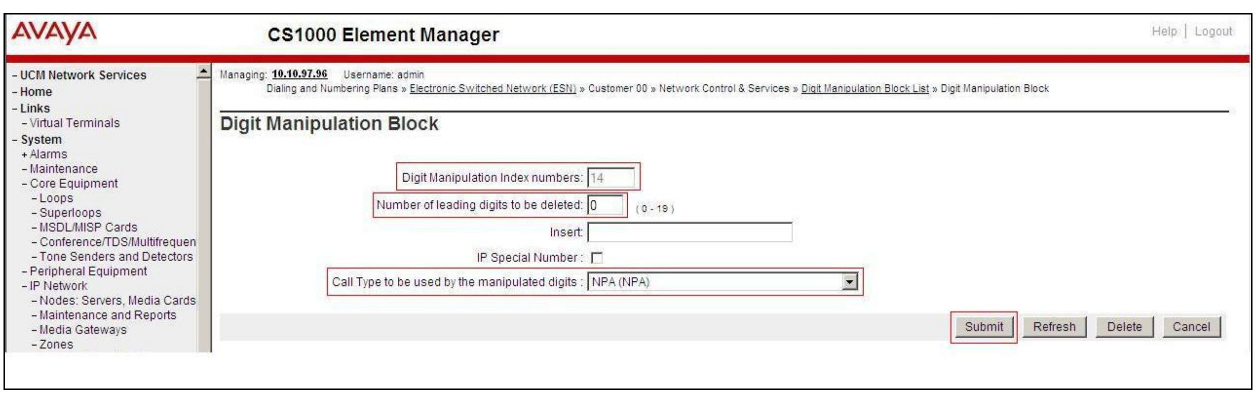


Figure 37 – DMI_14 Configuration Details

Add DMI_15: Enter **1** for the **Number of leading digits to be Deleted** (Del) field and select **NPA (NPA)** for the **Call Type to be used by the manipulated digits** field and click on the **Submit** button as shown in **Figure 38**.

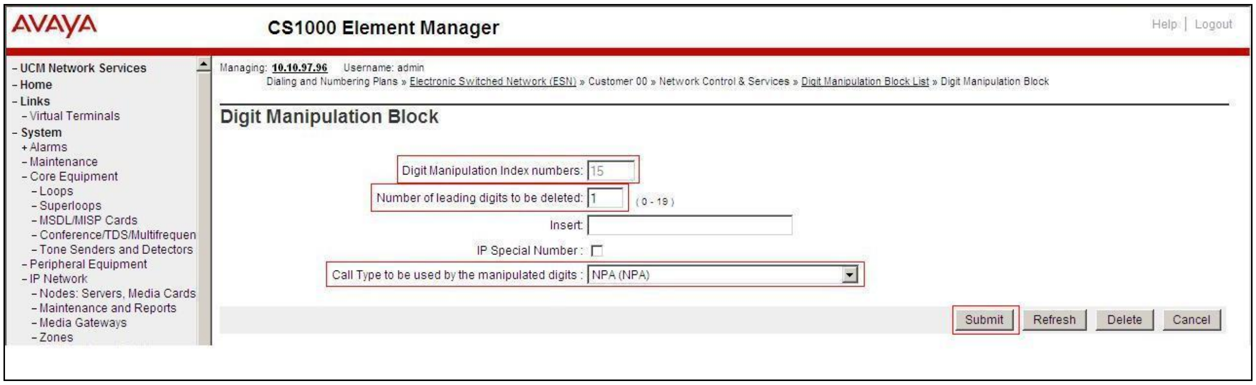


Figure 38 – DMI_15 Configuration Details

5.6.5. Route List Block (RLB) (RLB 14)

This session shows how to add a RLB associated with the DMI created in **Section 5.6.4**. Select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen. Select **Route List Block** (RLB) as shown in **Figure 34**.

Select an available value in the textbox for the **route list index** (in this case 14) and click on **to Add** button as shown in **Figure 39**.

Figure 39 – Add a Route List Block.

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

- **Digit Manipulation Index**: 14 (created in **Section 5.6.4**)
- **Incoming CLID Table**: 0 (created in **Section 5.5.7**)
- **Route number** 100 (created in **Section 5.5.5**)

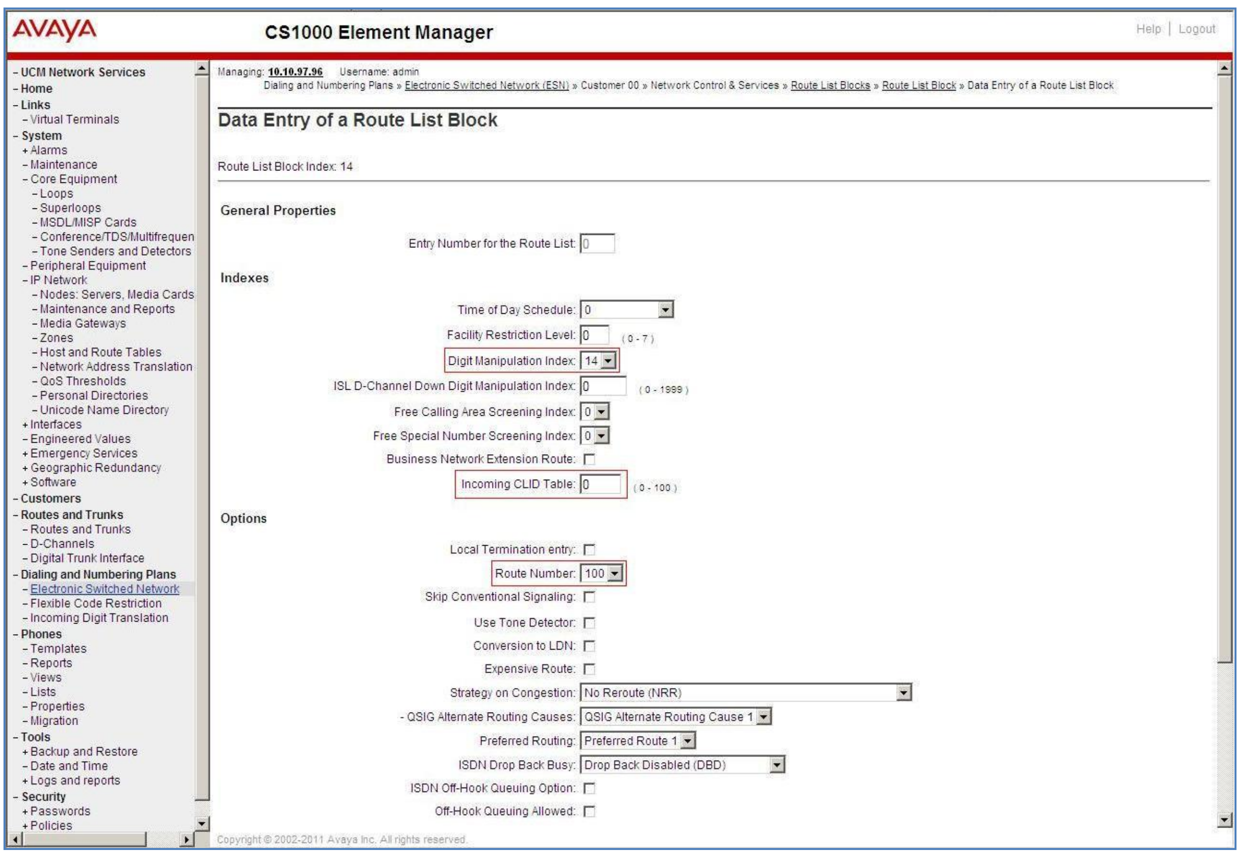


Figure 40 – RLB_14 Route List Block Configuration Details

5.6.6. Route List Block (RLB) (RLB 15)

Select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen. Select **Route List Block** (RLB) as shown in **Figure 34**. Select an available value in the textbox for the **route list block index** (in this case 15) and click on the **to Add** button as shown in **Figure 39**.
Enter the following values for the specified fields, and retain the default values for the remaining fields (**Figure 41**). Scroll down to the bottom of the screen, and click on the **Submit** button.

- **Digit Manipulation Index** (DMI): 15 (created in **Section 5.6.4**)
- **Incoming CLID Table**: 0 (created in **Section 5.5.7**)
- **Route number** (ROUT) : 100 (created in **Section 5.5.5**)

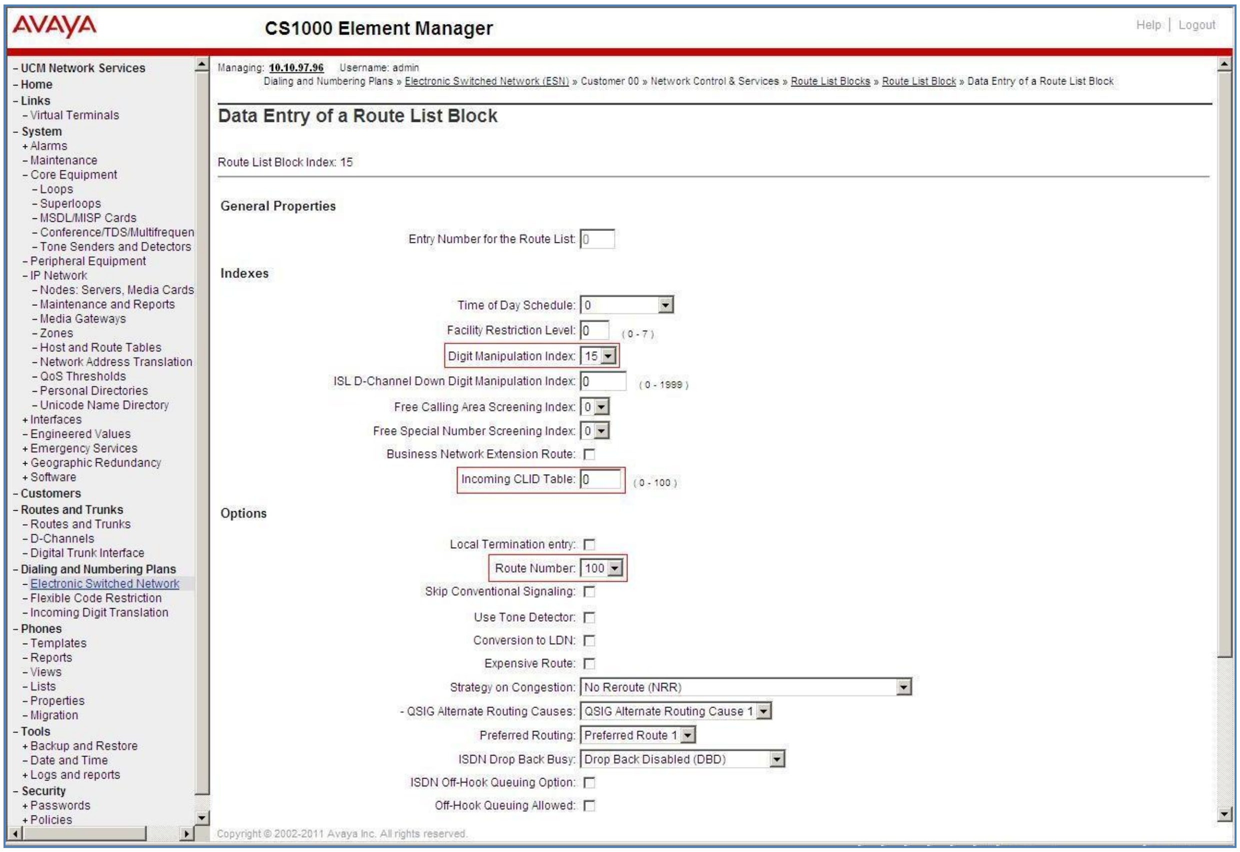


Figure 41 – RLB_15 Route List Block Configuration Details

5.6.7. Inbound Call – Incoming Digit Translation Configuration

This section describes the steps for receiving the calls from PSTN via the TELUS system. Select **Dialing and Numbering Plans → Incoming Digit Translation** from the left pane to display the **Incoming Digit Translation** screen. Click on the **Edit IDC** button as shown in **Figure 42**.

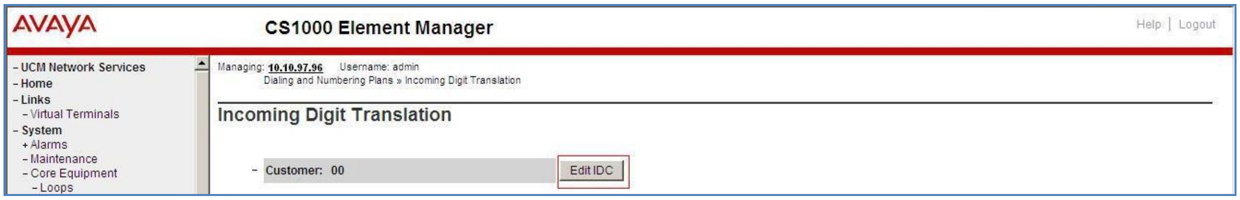


Figure 42 – Incoming Digit Translation

Click on the **New DCNO** to create the digit translation mechanism. In this example, Digit Conversion Tree Number 1 has been created as shown in **Figure 43**.

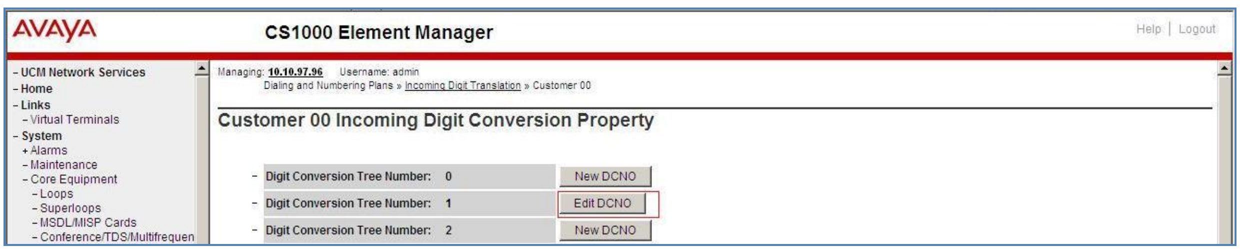


Figure 43 – Incoming Digit Conversion Property

Detail configuration of the Digit Conversion Tree Configuration is shown in **Figure 44**. The **Incoming Digits** can be added to map to the Converted Digits which would be the Communication Server 1000 system phones DN. This **DCNO** has been assigned to route 100 as shown in **Figure 26** and **27**.

In the following configuration, the incoming call from PSTN with the prefix 403692xxxx will be translated to DN xxxx. The DID number 4036929470 is translated to 1700 for Voicemail accessing purpose.

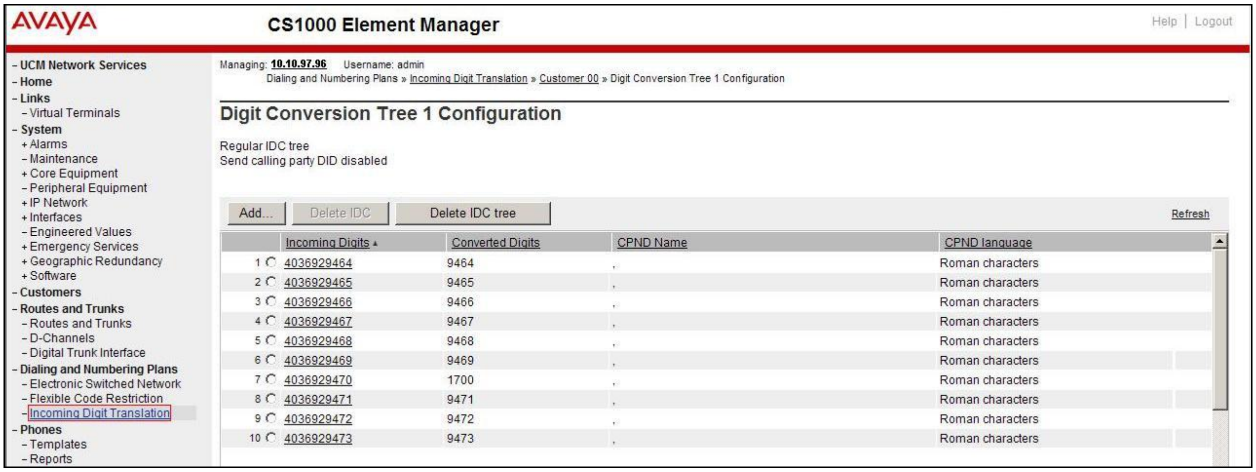


Figure 44 – Digit Conversion Tree

5.6.8. Outbound Call - Special Number Configuration

There are special numbers which have been configured to be used for this testing such as: 0, 011, 411, 911 and so on.

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

Enter SPN number and then click on **to Add** button. **Figure 45** shows all the special number used for this testing.

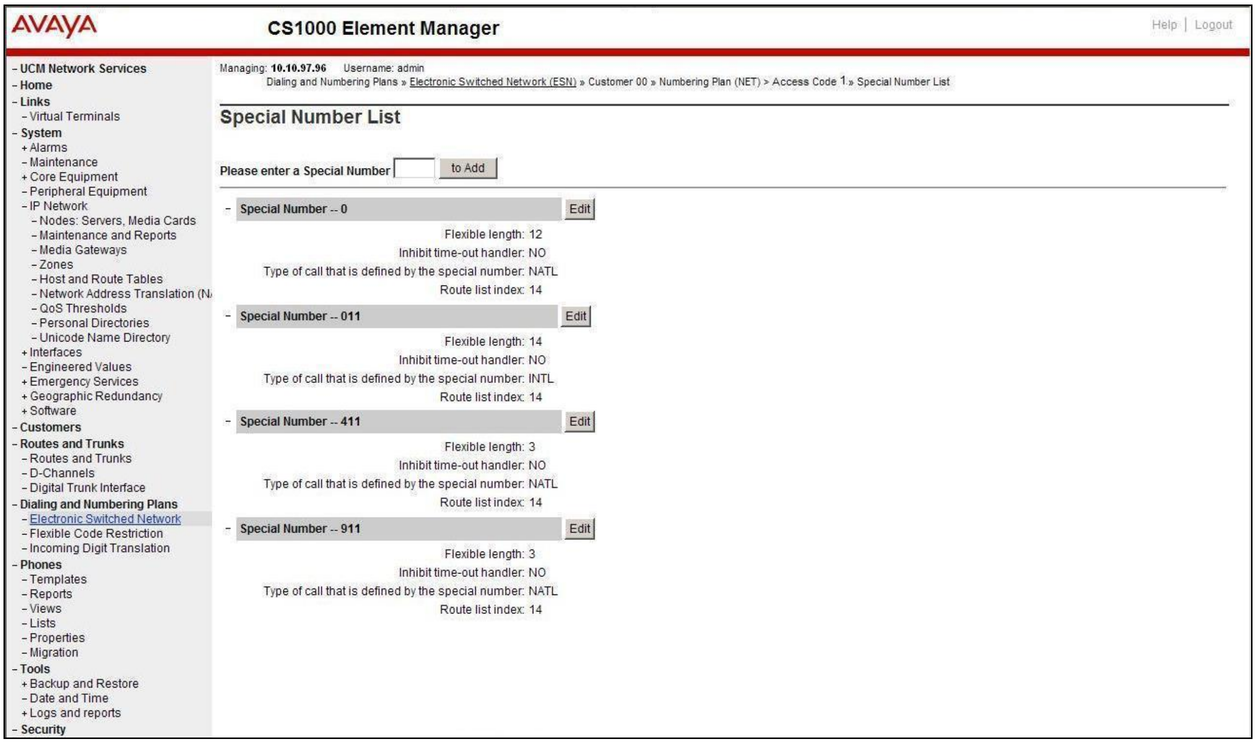


Figure 45 – Add a SPN.

5.6.9. Outbound Call - Numbering Plan Area (NPA)

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

Select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen. Select **Numbering Plan Area Code** (NPA) as shown in **Figure 34**.
Enter the area code desired in the textbox and click on the **to Add** button. The 1403,1416, 1604, 1613, 1647, 1780 and 1800 area codes were used in this configuration as shown in **Figure 46**.

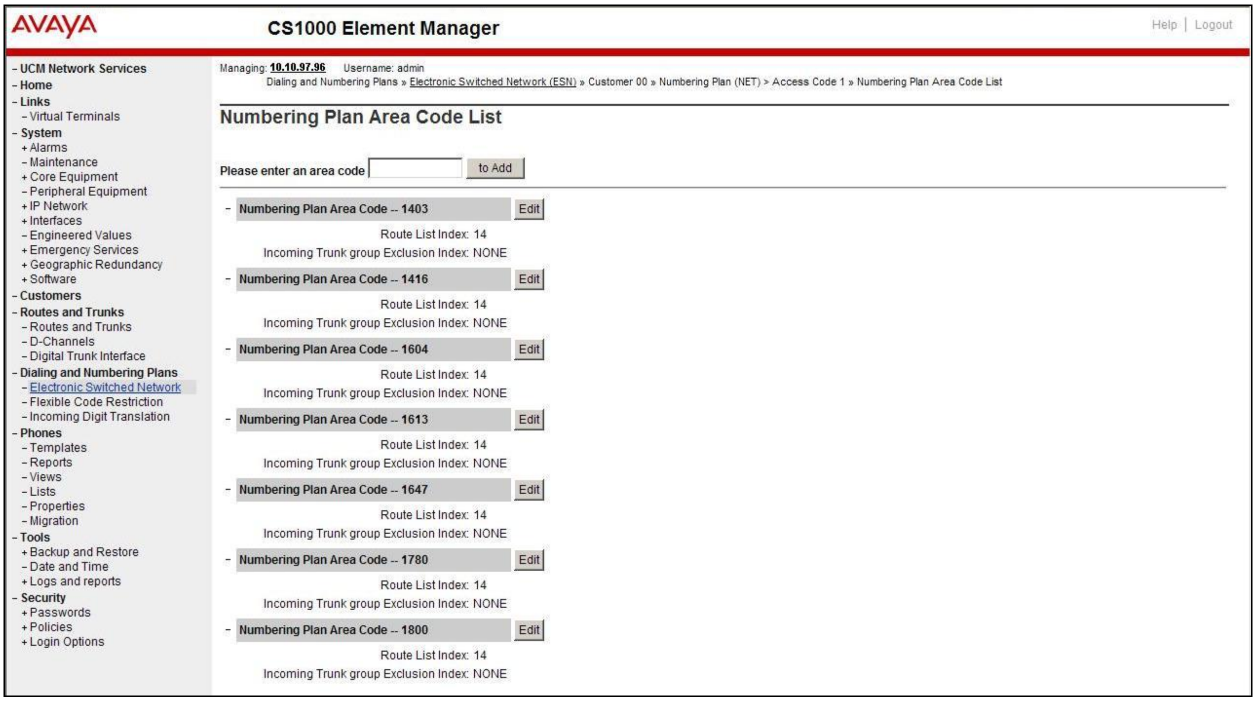


Figure 46 – Numbering Plan Area Code List

5.7. Administer Phone

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

5.7.1. Phone creation

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

```
REQ: prt
TYPE: 2002p2
TN   96 0 0 2
DATE
PAGE
DES
MODEL_NAME
EMULATED

DES  2002P2
TN   96 0 00 02  VIRTUAL
TYPE 2002P2
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 00010
CUR_ZONE 00010
MRT
ERL  12345
ECL  0
FDN
TGAR 0
LDN  NO
NCOS 7
SGRP 0
RNPG 0
SCI  0
SSU
LNRS 16
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS  UNR FBD WTA LPR MTD FND HTD TDD CRPD
     MWD LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LNA CNDA
     CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
     ICDD CDMD LLCN MCTD CLBD AUTU
     GPUD DPUD DNDD CFXD ARHD CLTD ASCD
```

```
    CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
    UDI RCC HBTD AHD IPND  DDGA NAMA MIND PRSD NRWD NRCD NROD
    DRDD EXR0
    USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
    FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD
    MSNV FRA  PKCH MWTD DVLD CROD ELCD
CPND_LANG ENG
HUNT
PLEV 02
PUID
UPWD
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY  00 SCR 9464 0     MARP
      CPND
        CPND_LANG ROMAN
          NAME Carrier1
          XPLN 13
          DISPLAY_FMT FIRST,LAST
    01
    02
<Text removed for brevity>
```

5.7.2. Enable Privacy for Phone

In this section, it shows how to enable Privacy for a phone by changing its class of service (CLS). By modifying the configuration of the phone created in **Section 5.7.1**, the display of the outbound call will be changed appropriately.
To hide the display number, set CLS to **ddgd**. Communication Server 1000 will include “Privacy:id” in the SIP message header before sending it to the Service Provider.

```
>ld 11
REQ: chg
TYPE: 2002p2
TN   96 0 0 2
ECHG yes
ITEM cls ddgd
…
```

To allow display number, set CLS to **ddga**. Communication Server 1000 will not send the Privacy header to the Service Provider.

```
>ld 11
REQ: chg
TYPE: 2002p2
TN   96 0 0 2
ECHG yes
ITEM cls ddga
…
```

5.7.3. Enable Call Forward for Phone

In this section, it shows how to configure the Call Forward feature at the system and phone level. Select **Customer → 00 → Call Redirection**. The Call Redirection page is shown in **Figure 47**.

- **Total redirection count limit**: **0** (unlimited)
- **Call Forward: Originating**
- **Number of normal ring cycle of CFNA: 4**
- Click **Save** to save the configuration.

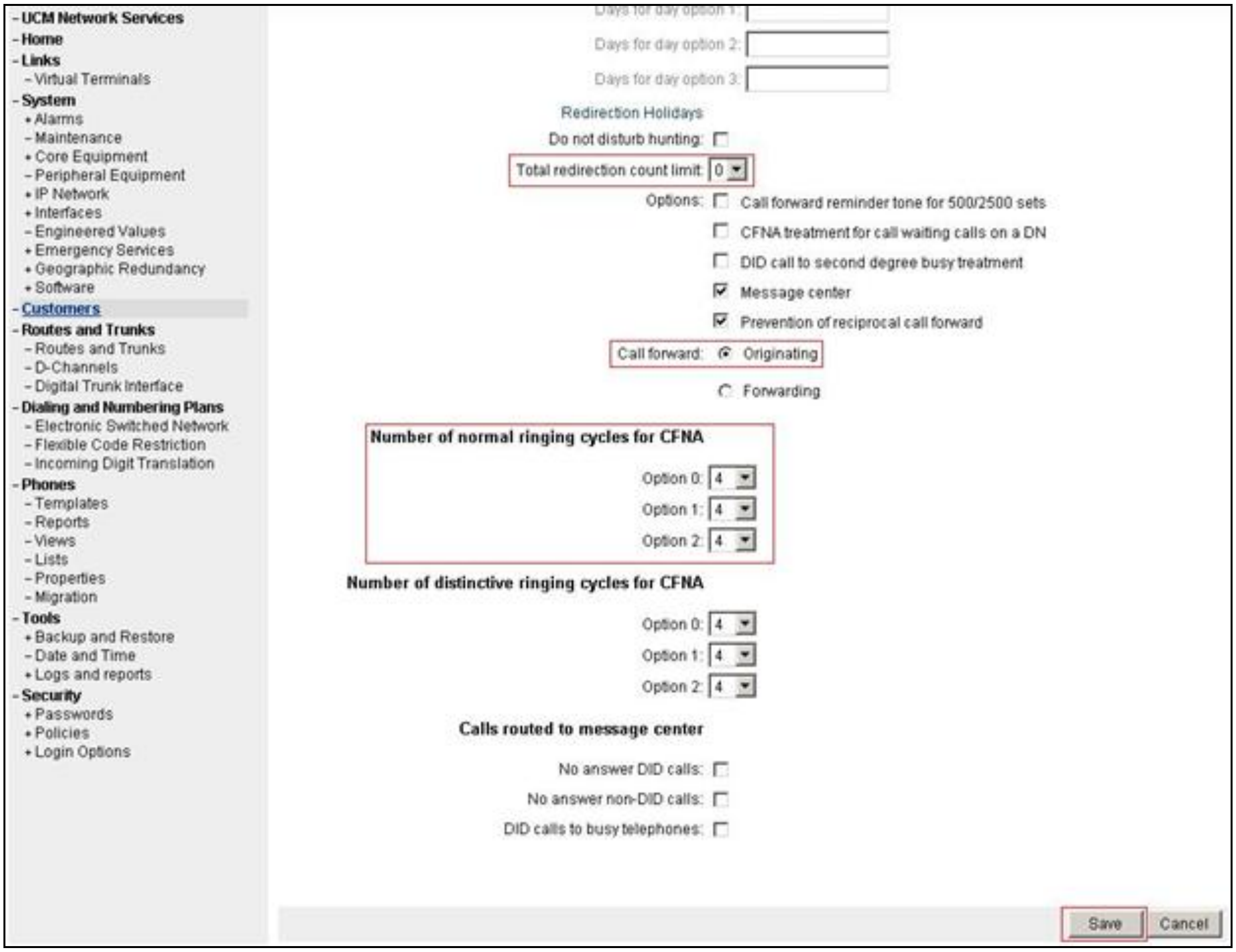


Figure 47 – Call Redirection

To enable **Call Forward All Call** (**CFAC**) for a phone over a trunk, use **LD 11**, change its CLS to **CXFA**, **SFA** then program the forward number on the phone set. Following is the configuration of a phone that has **CFAC** enabled with forwarding number 616139675205.

```
REQ: prt
TYPE: 2007
TN   96 0 0 4
```

```
DATE
PAGE
DES
MODEL_NAME
EMULATED

DES  2007
TN   96 0 00 04  VIRTUAL
TYPE 2007
…
CLS  UNR FBA WTA LPR MTD FNA HTA TDD HFD CRPD
    MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
    POD SLKD CCSD SWD LNA CNDA
    CFTD SFA MRD DDV CNID CDCA MSID DAPA BFED RCBD
    ICDA CDMA LLCN MCTD CLBD AUTU
    GPUD DPUD DNDD CFXA ARHD CLTD ASCD
…
    19 CFW 16  616139675205
```

To enable **Call Forward Busy (CFB)** for phone over trunk by using **LD 11**, change its **CLS** to **FBA, HTA, SFA** then program the forward number as is **HUNT**. Following is the configuration of a phone has **CFB** enabled with forward number is 616139675205.

```
REQ: prt
TYPE: 2007
TN   96 0 0 4
DATE
PAGE
DES
MODEL_NAME
EMULATED

DES  2007
TN   96 0 00 04  VIRTUAL
TYPE 2007
…
CLS  UNR FBA WTA LPR MTD FNA HTA TDD HFD CRPD
    MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
    POD SLKD CCSD SWD LNA CNDA
    CFTD SFA MRD DDV CNID CDCA MSID DAPA BFED RCBD
…
FDN 616139675205
HUNT 616139675205
```

To enable **Call Forward No Answer (CFNA)** for a phone over a trunk by using **LD 11**, change its **CLS** to **FNA, SFA** then program the forward number as **FDN**. Following is the configuration of a phone that has CFNA enabled with forward number 616139675205.

```
REQ: prt
TYPE: 2007
TN   96 0 0 4
```

```
DATE
PAGE
DES
MODEL_NAME
EMULATED

DES  2007
TN   96 0 00 04  VIRTUAL
TYPE 2007
…
FDN  616139675205
HUNT  616139675205

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

5.7.4. Enable Call Waiting for Phone

In this section, it shows how to configure Call Waiting feature at phone level.
Log in to the Call Server CLI (please refer to **Section 5.1.2** for more detail), configure Call Waiting feature for phone by using **LD 11** to change **CLS** to **HTD**, **SWA** and adding a **CWT** key.

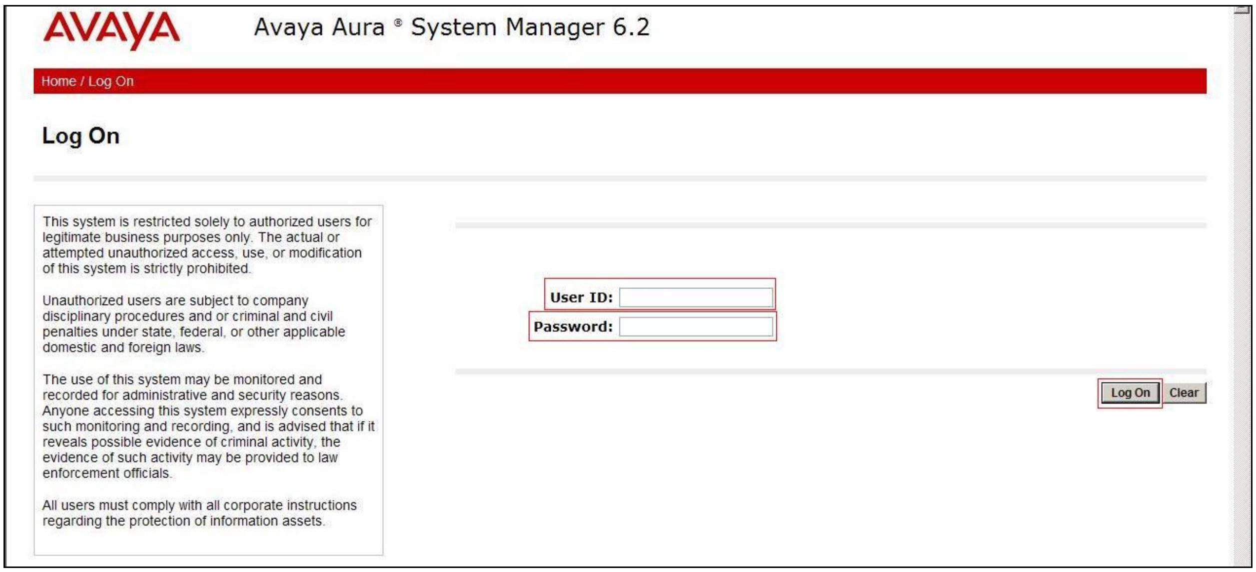
```
REQ: prt
TYPE: 2002p2

TN   96 0 0 2
DATE
PAGE
DES
MODEL_NAME
EMULATED
KEM_RANGE

DES  2002P2
TN   96 0 00 02  VIRTUAL
TYPE 2002P2
…
CLS  UNR FBD WTA LPR MTD FNA HTD TDD HFD CRPD
    MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
    POD SLKD CCSD SWA LNA CNDA
…
KEY  00 SCR 9464 0     MARP
       CPND
         CPND_LANG ROMAN
           NAME Carrier1
           XPLN 13
           DISPLAY_FMT FIRST,LAST
    01 CWT
…
```

6. Configure Avaya Aura® Session Manager

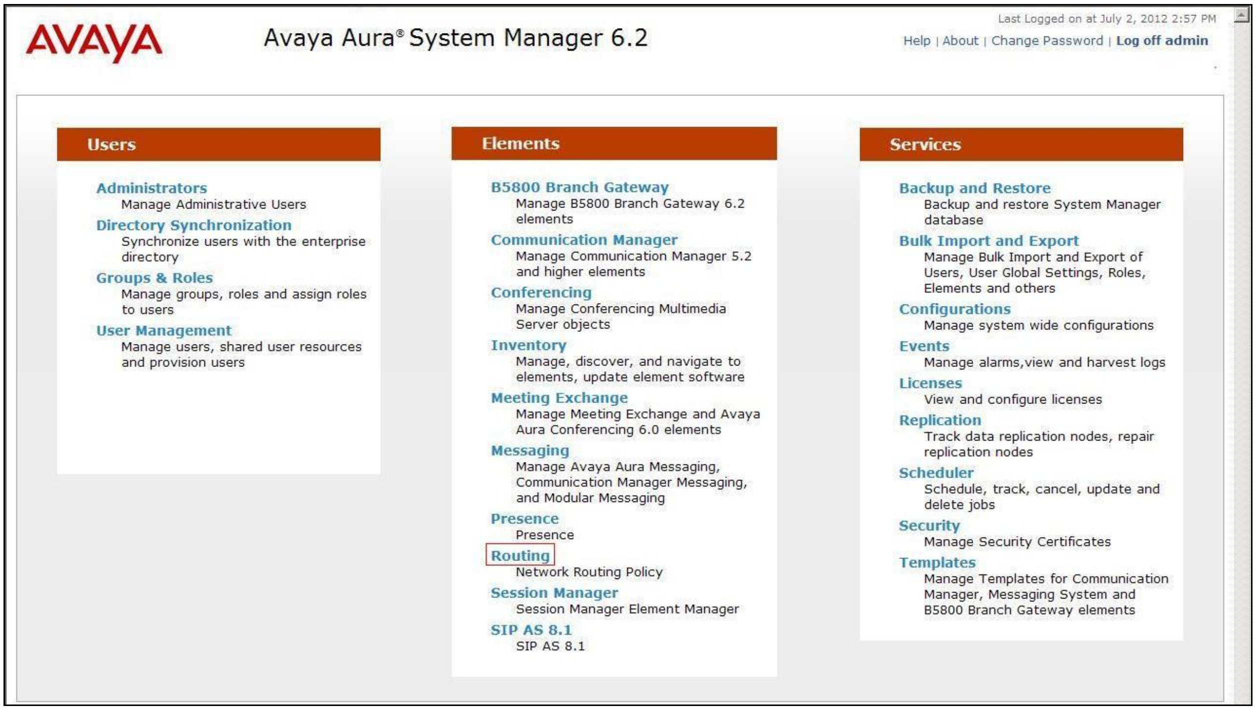
This section illustrates relevant aspects of the Session Manager configuration used in the verification of these Application Notes.

Note: The following sections assume that Session Manager and System Manager have been installed and that network connectivity exists between the two. For more information on Session Manager see **Section 10** of these Application Notes.

Session Manager is managed via System Manager. Using a web browser, access https://<ip-addr of System Manager>/SMGR. In the **Log On** screen, enter appropriate **User ID** and **Password** and press the **Log On** button



Once logged in, a Home Screen is displayed as below:



When **Routing** is selected, the right side of Routing outlines a series of steps.

The sub-sections that follow are in the same order as the steps outlined under **Introduction to Network Routing Policy (NRP)** in the abridged screen shown below. In these Application Notes, all these steps are illustrated with the exception of Step 9, since Regular Expressions were not used.

Introduction to Network Routing Policy

Network Routing Policy consists of several routing applications like "Domains", "Locations", "SIP Entities", etc.

The recommended order to use the routing applications (that means the overall routing workflow) to configure your network configuration is as follows:

Step 1: Create "Domains" of type SIP (other routing applications are referring domains of type SIP).

Step 2: Create "Locations"

Step 3: Create "Adaptations"

Step 4: Create "SIP Entities"

- SIP Entities that are used as "Outbound Proxies" e.g. a certain "Gateway" or "SIP Trunk"
- Create all "other SIP Entities" (Session Manager, CM, SIP/PSTN Gateways, SIP Trunks)
- Assign the appropriate "Locations", "Adaptations" and "Outbound Proxies"

Step 5: Create the "Entity Links"

- Between Session Managers
- Between Session Managers and "other SIP Entities"

Step 6: Create "Time Ranges"

- Align with the tariff information received from the Service Providers

Step 7: Create "Routing Policies"

- Assign the appropriate "Routing Destination" and "Time Of Day"

(Time Of Day = assign the appropriate "Time Range" and define the "Ranking")

Step 8: Create "Dial Patterns"

- Assign the appropriate "Locations" and "Routing Policies" to the "Dial Patterns"

Step 9: Create "Regular Expressions"

- Assign the appropriate "Routing Policies" to the "Regular Expressions"

Each "Routing Policy" defines the "Routing Destination" (which is a "SIP Entity") as well as the "Time of Day" and its associated "Ranking".

IMPORTANT: the appropriate dial patterns are defined and assigned afterwards with the help of the routing application "Dial patterns". That's why this overall routing workflow can be interpreted as

"Dial Pattern driven approach to define Routing Policies"

That means (with regard to steps listed above):

Step 7: "Routing Polices" are defined

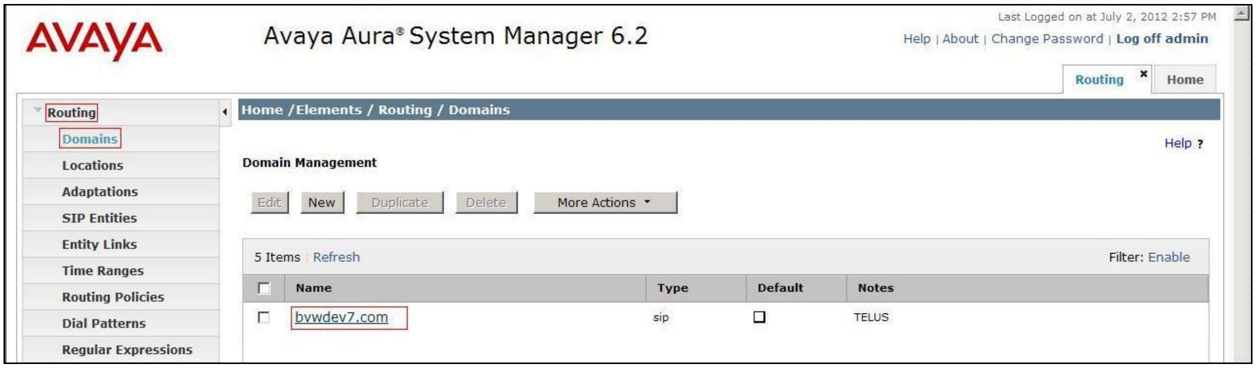
Step 8: "Dial Patterns" are defined and assigned to "Routing Policies" and "Locations" (one step)

Step 9: "Regular Expressions" are defined and assigned to "Routing Policies" (one step)

6.1. Configure Domains

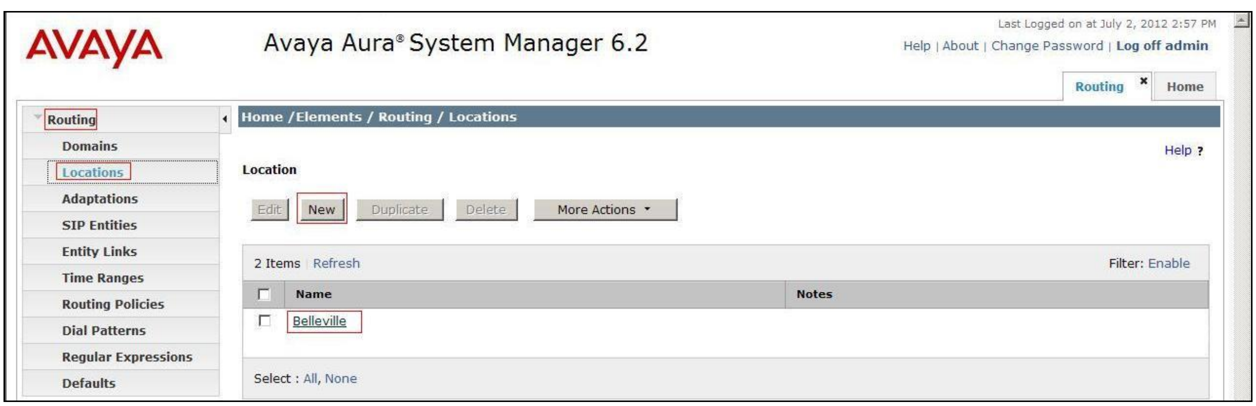
To add SIP domains that will be used with Session Manager, select **Routing → Domains**. Click the **New** button to add a new SIP domain entry. Click the Commit button after changes are completed.

The following screen shows the list of configured SIP domains. The domain **bvwdev7.com** is not known to the TELUS service. The domain name should match the one used in the **SIP domain name** described in **Section 5.5.2**.

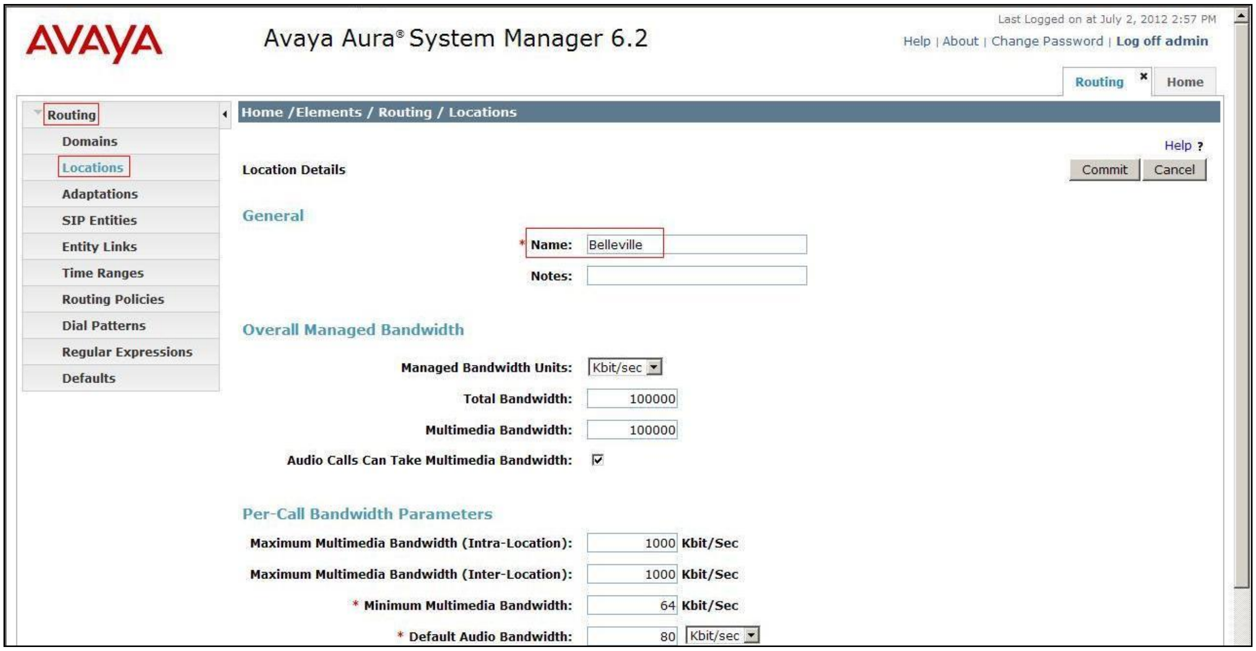


6.2. Configure Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside, for the purposes of bandwidth management. One location is added to the sample configuration for the enterprise SIP entities. To add locations, select **Routing → Locations**. The following screen shows an abridged list of configured locations. Click on the checkbox corresponding to the name of a location and the **New** button to add a location. Click the Commit button after changes are completed. Assigning unique locations can allow Session Manager to perform location-based routing, bandwidth management, and call admission control.

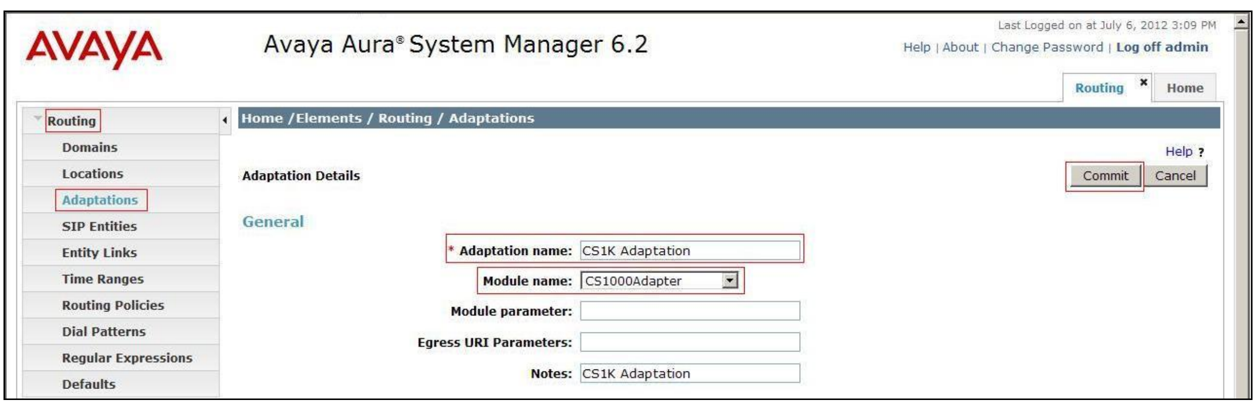


The following screen shows the location details for the location named **Belleville**, corresponding to the Avaya SBCE. Later, the location with name Belleville will be assigned to the corresponding SIP Entity.

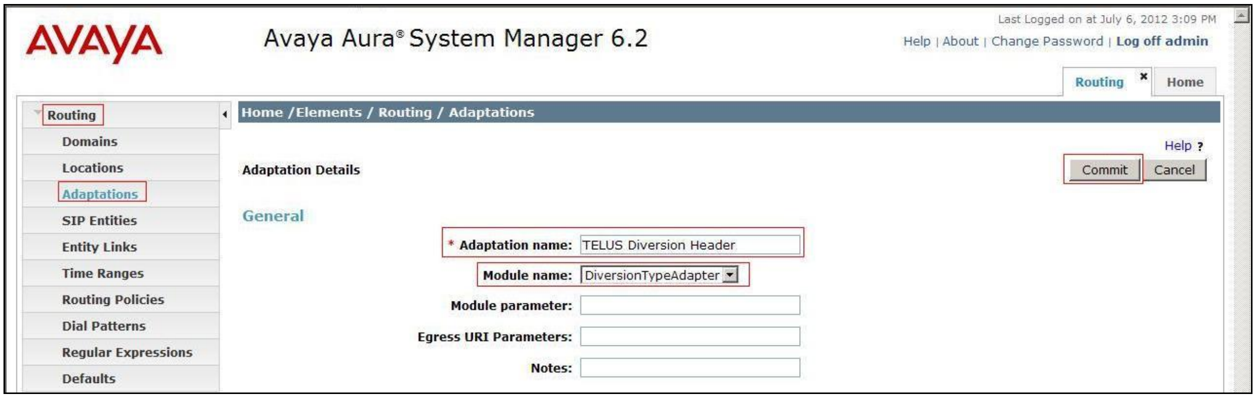


6.3. Configure Adaptations

Adaptation is configured to format the History Info on Communication Server 1000 to be compatible with Avaya History Info form. In order to add a new adaptation, select **Routing → Adaptations**. Click the New button to add an adaptation. Enter an appropriate **Adaptation name** to identify the adaptation. Select **CS1000Adapter** from the **Module name** drop-down menu. Click the **Commit** button after changes are completed.



Adaptation is configured to convert the History Info to Diversion Header. In order to add a new adaptation, select **Routing → Adaptations**. Click the New button to add an adaptation. Enter an appropriate **Adaptation name** to identify the adaptation. Select **DiversionTypeAdapter** from the **Module name** drop-down menu. Click the **Commit** button after changes are completed.



6.4. Configure SIP Entities

A SIP Entity must be added for each SIP-based telephony system supported by a SIP connection to the Session Manager. To add a SIP Entity, select **Routing → SIP Entities** and then click on the **New** button. The following will need to be entered for each SIP Entity.

Under **General:**

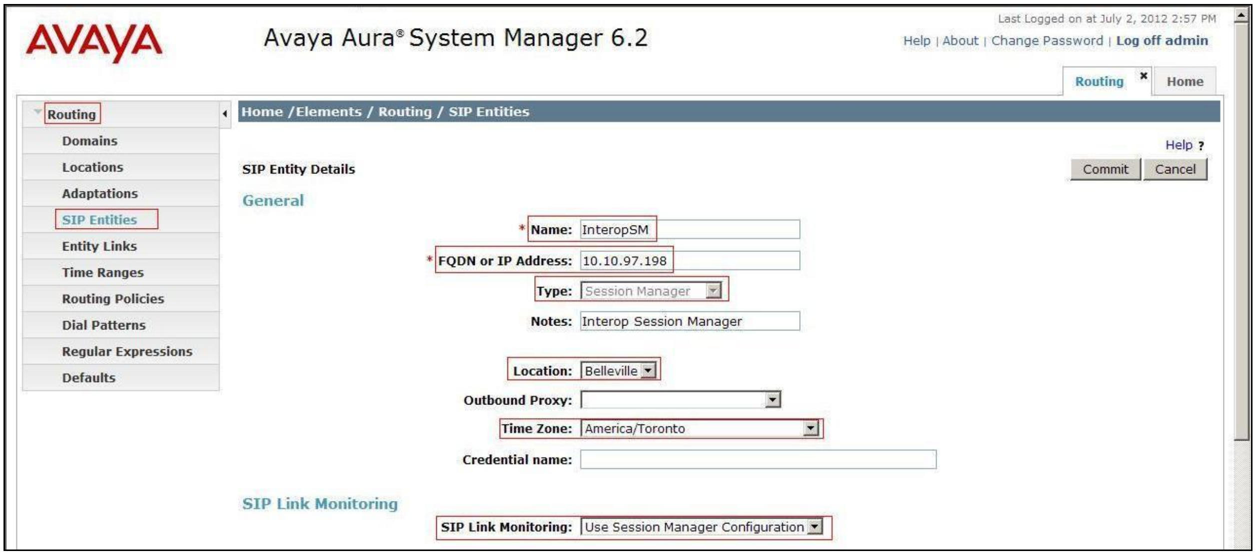
- In the **Name** field enter an informative name
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signaling interface on the connecting system.
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, ***Other*** for the Communication Server 1000 and the Avaya SBCE
- In the **Location** field select the appropriate location (configured in **Section 6.2**) from the drop down menu
- In the **Time Zone** field enter the time zone for the SIP Entity

In this configuration there are three SIP Entities.

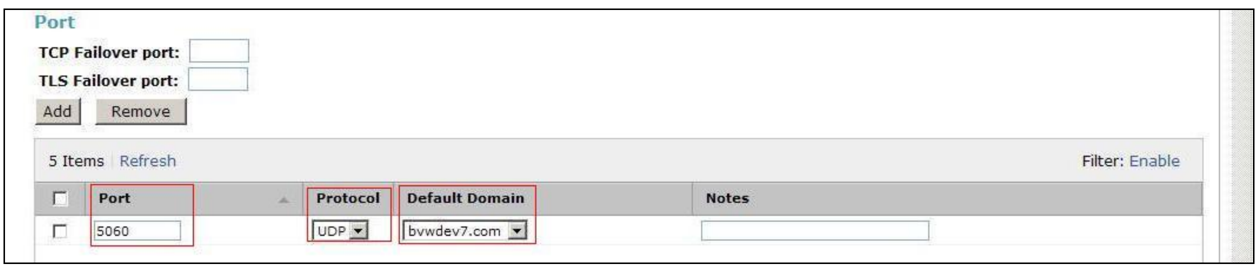
- Session Manager SIP Entity
- Avaya SBCE SIP Entity
- Communication Server 1000 SIP Entity

6.4.1. Configure Avaya Aura® Session Manager SIP Entity

The following screens show the SIP entity for Session Manager named **InteropSM**. The **IP Address** field is set to the IP address **10.10.97.198** of the Session Manager SIP signaling interface.



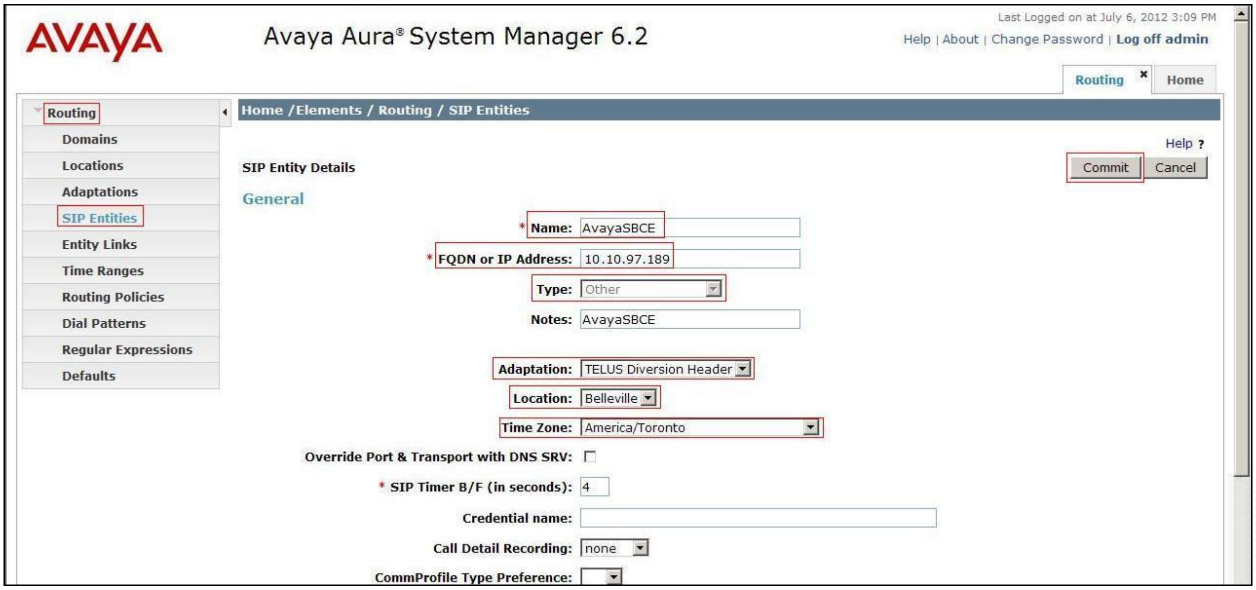
Click the **Add** button under the Port section to configure a new port. **Protocol UDP** is used in the sample configuration for improved visibility during testing. **Port** is **5060, Protocol** is **UDP** and **Default Domain** is **bvwdev7.com**.
Click the **Commit** button (not shown) after changes are completed.



6.4.2. Configure Avaya SBCE SIP Entity

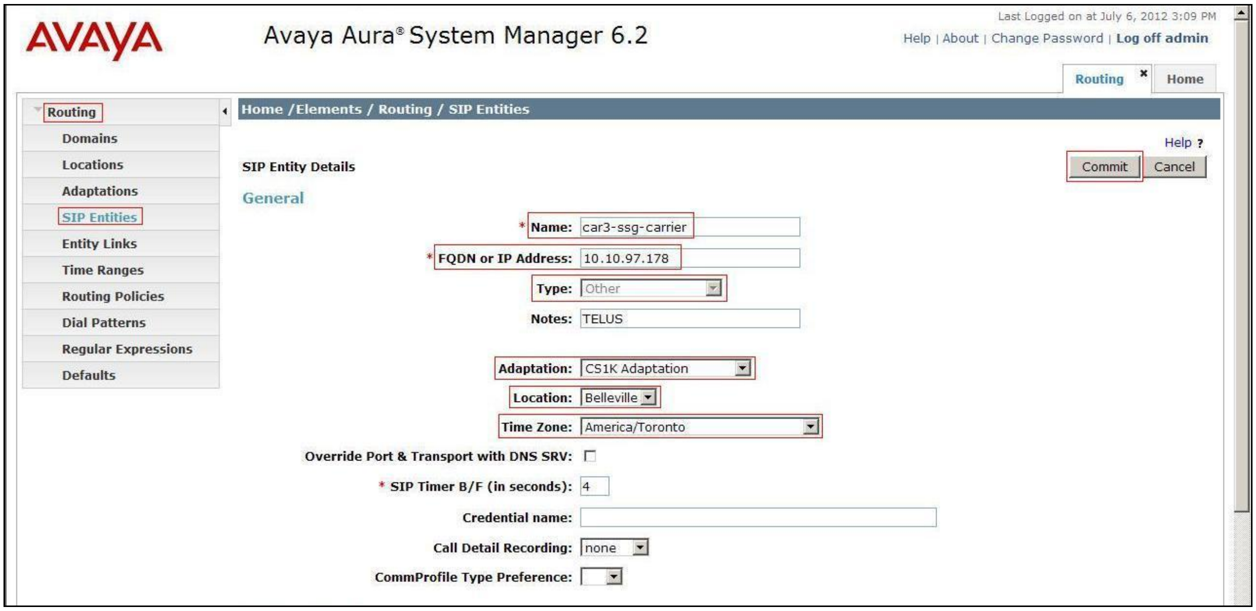
The following screen shows the **SIP Entity Details** for the Avaya SBCE named **AvayaSBCE**. The **Adaptation: TELUS Diversion Header** is in use. The **IP Address** field is configured with the Avaya SBCE inside IP Address (**10.10.97.189**). Choose **Type** as **Other** and **Location as Belleville**. Set **Time Zone** as **America/Toronto**.

Click **Commit** to save the configuration.



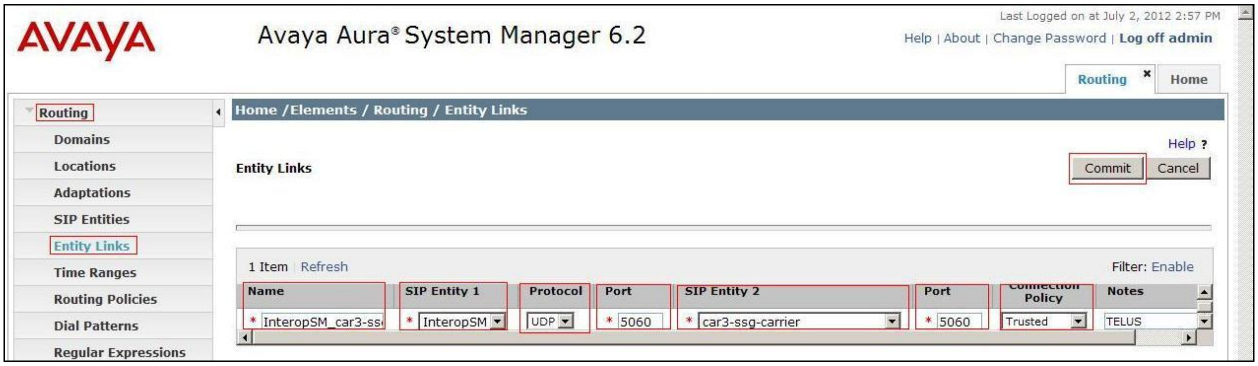
6.4.3. Configure Communication Server 1000 SIP Entity

The following screen shows a portion of the **SIP Entity Details** corresponding to a Communication Server SIP Entity named **car3-ssg-carrier**. The **Adaptation**: **CS1K Adaptation** is in use. The **IP Address** field contains the IP Address of Node ID **10.10.97.178**. Choose **Type** as **Other** and **Location as Belleville**. Set **Time Zone** as **America/Toronto**
Click **Commit** to save the configuration.

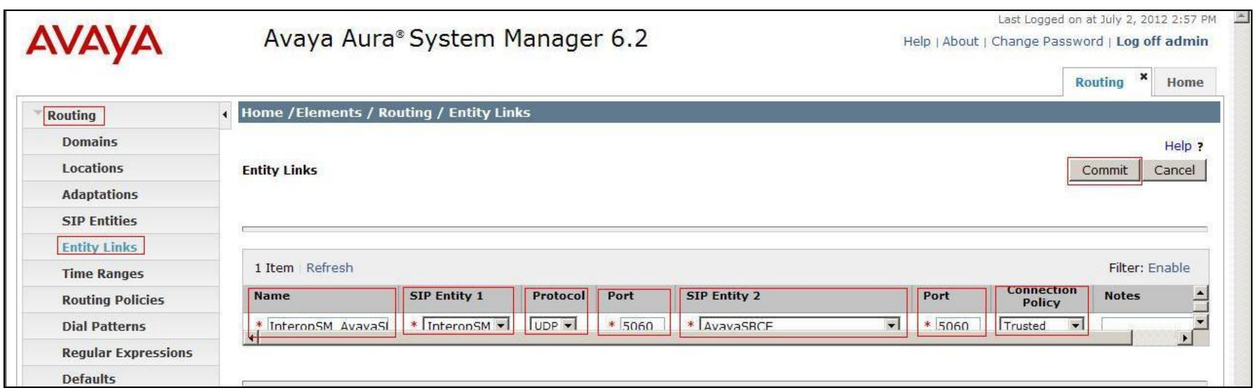


6.5. Configure Entity Links

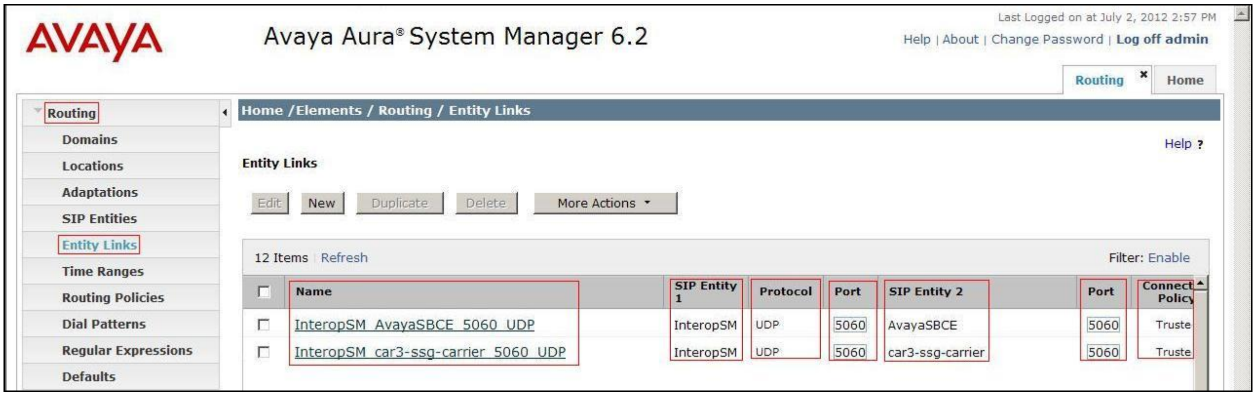
A SIP trunk between a Session Manager and another system is described by an Entity Link. To add an Entity Link, select **Routing → Entity Links**. Click the **New** button to add a link for Communication Server 1000. Assign an appropriate **Name**, and select the Session Manager entity as **SIP Entity 1,** and the Communication Server 1000 entity as **SIP Entity 2**. Assign the **Protocol** as **UDP**, select **Port 5060**, select **Connection Policy Trusted**, and click **Commit**.



Click the **New** button to add a link for the Avaya SBCE. Assign an appropriate **Name**, and select the Session Manager entity as **SIP Entity 1,** and the Avaya SBCE entity as **SIP Entity 2**. Assign the **Protocol** as **UDP**, select **Port 5060**, select **Connection Policy Trusted**, and click **Commit**.

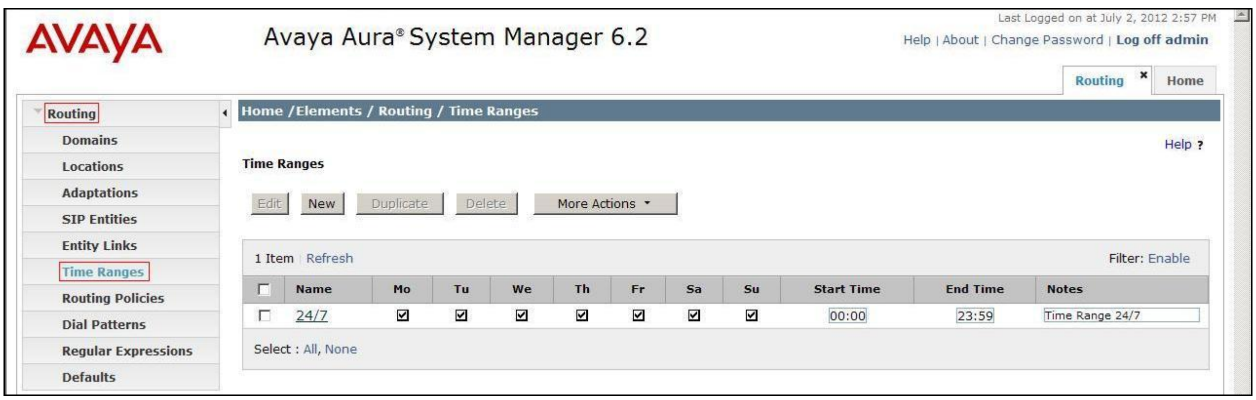


The following screen shows the list of configured links. Each of the links uses the entity named **InteropSM** as **SIP Entity 1**, and the appropriate entities, such as **car3-ssg-carrier**, **AvayaSBCE** for **SIP Entity 2**.



6.6. Configure Time Ranges

Time Ranges is configured for time-based-routing. In order to add a Time Ranges, select **Routing → Time Ranges** and then click **New** button. The Routing Policies shown subsequently will use the 24/7 range since time-based routing was not the focus of these Application Notes.



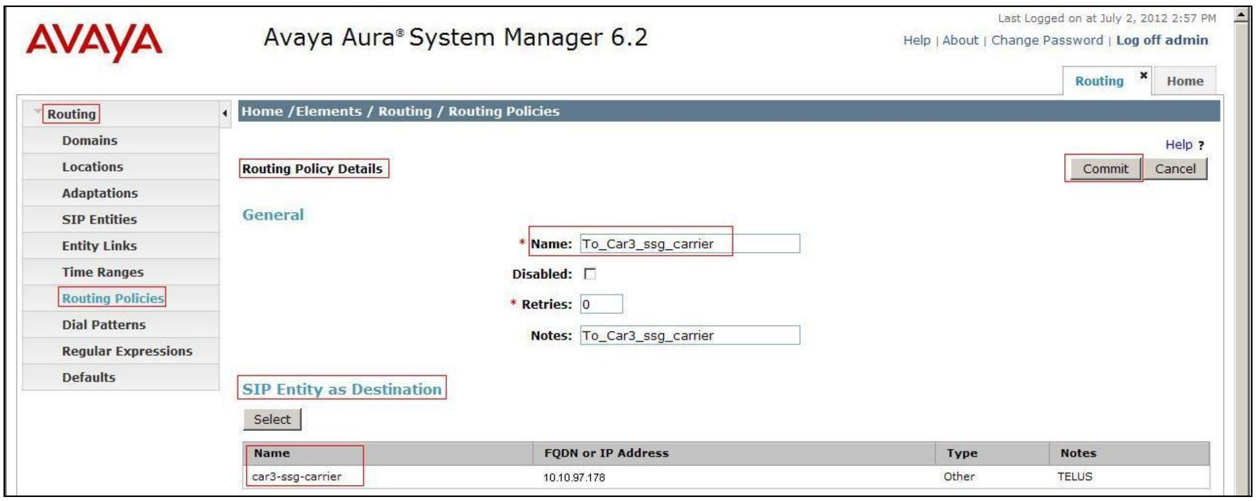
6.7. Configure Routing Policies

Routing policies must be created to direct how calls will be routed to a system. To add a new routing policy, select **Routing → Routing Policies** and then click on the **New** button.

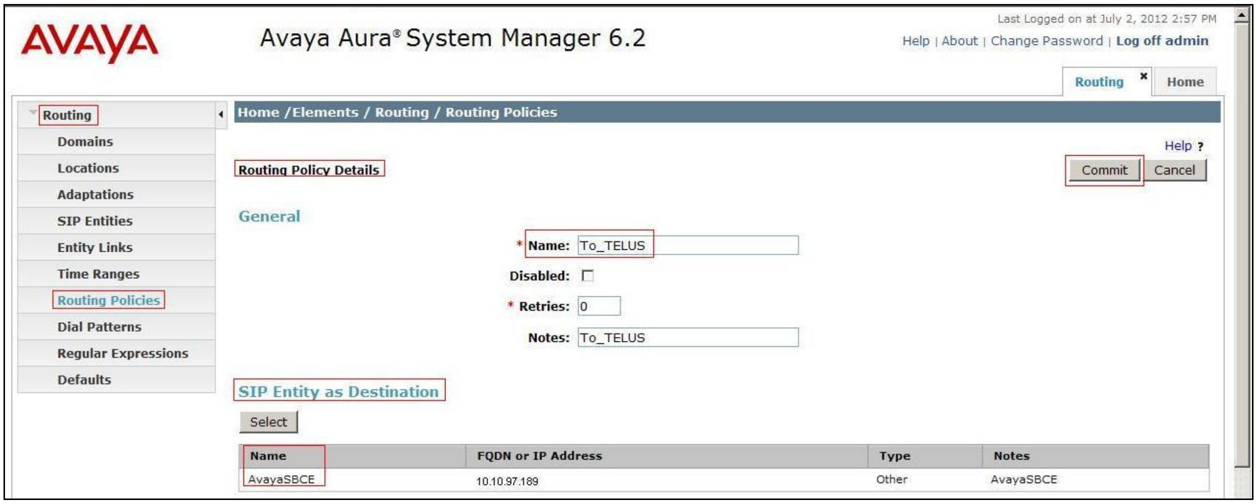
Under **General:**

- Enter an informative name in the **Name** field
- Under **SIP Entity as Destination**, click **Select**, and then select the appropriate SIP entity to which this routing policy applies

The following screen shows the **Routing Policy Details** for the policy named **To_Car3_ssg_carrier** associated with incoming PSTN calls from TELUS to Communication Server 1000. Observe the **SIP Entity as Destination** is the entity named **car3-ssg-carrier**.



The following screen shows the **Routing Policy Details** for the policy named **To_TELUS** associated with outgoing calls from Communication Server 1000 to the PSTN via TELUS through the Avaya SBCE. Observe the **SIP Entity as Destination** is the entity named **AvayaSBCE**



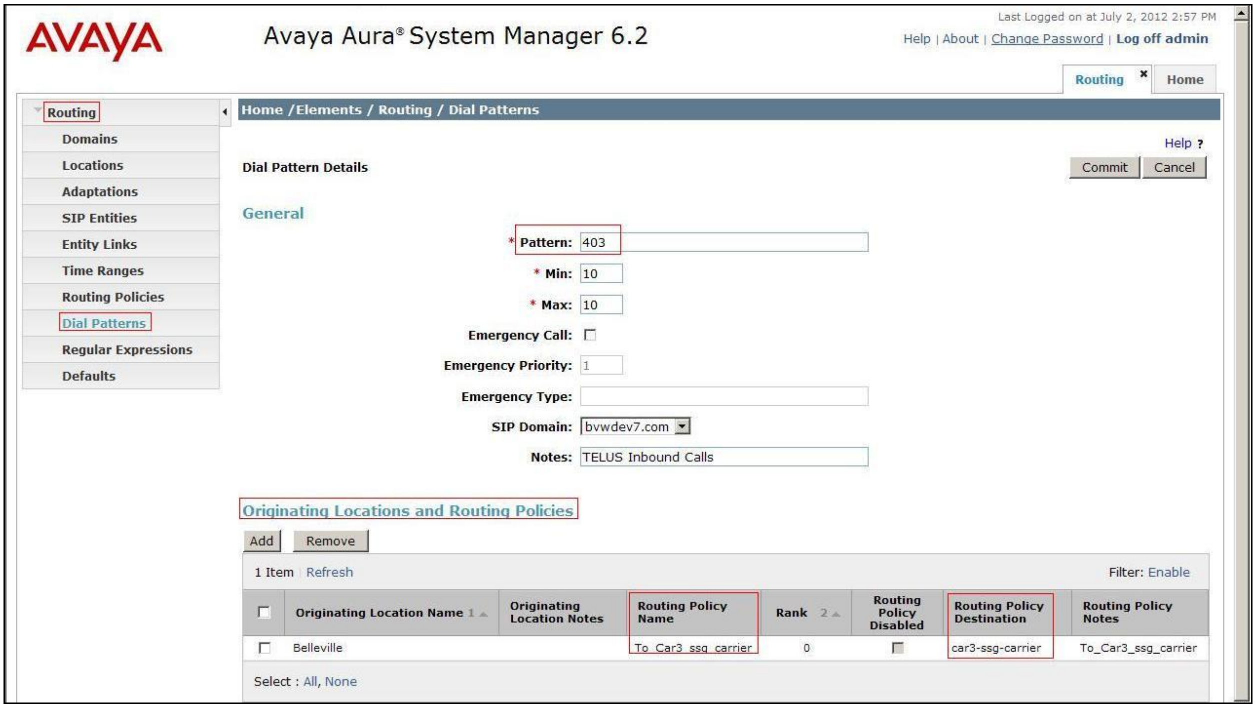
6.8. Configure Dial Patterns

A dial pattern must be defined to direct calls to the appropriate telephony system. To add a new dial pattern, select **Routing → Dial Patterns** and then click on the **New** button.

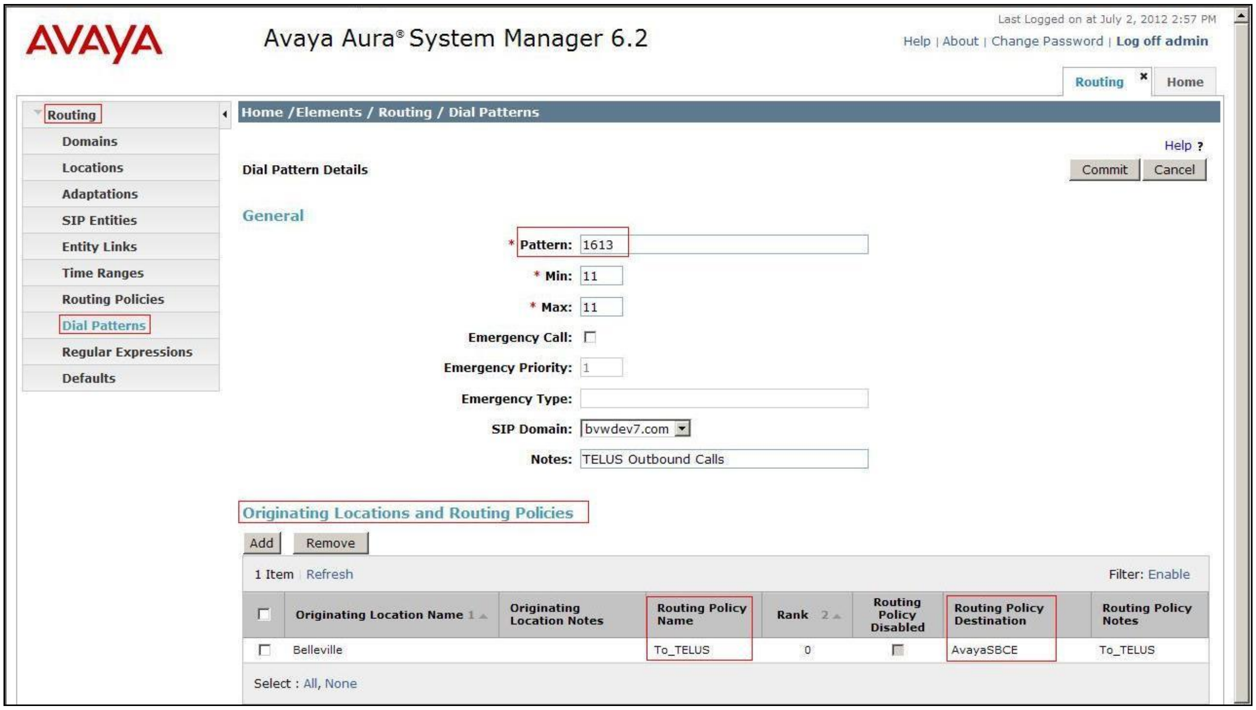
Under **General:**

- In the **Pattern** field, enter a dialed number or prefix to be matched
- In the **Min** field, enter the minimum length of the dialed number
- In the **Max** field, enter the maximum length of the dialed number
- In the **SIP Domain** field, select the domain configured in **Section 6.1**

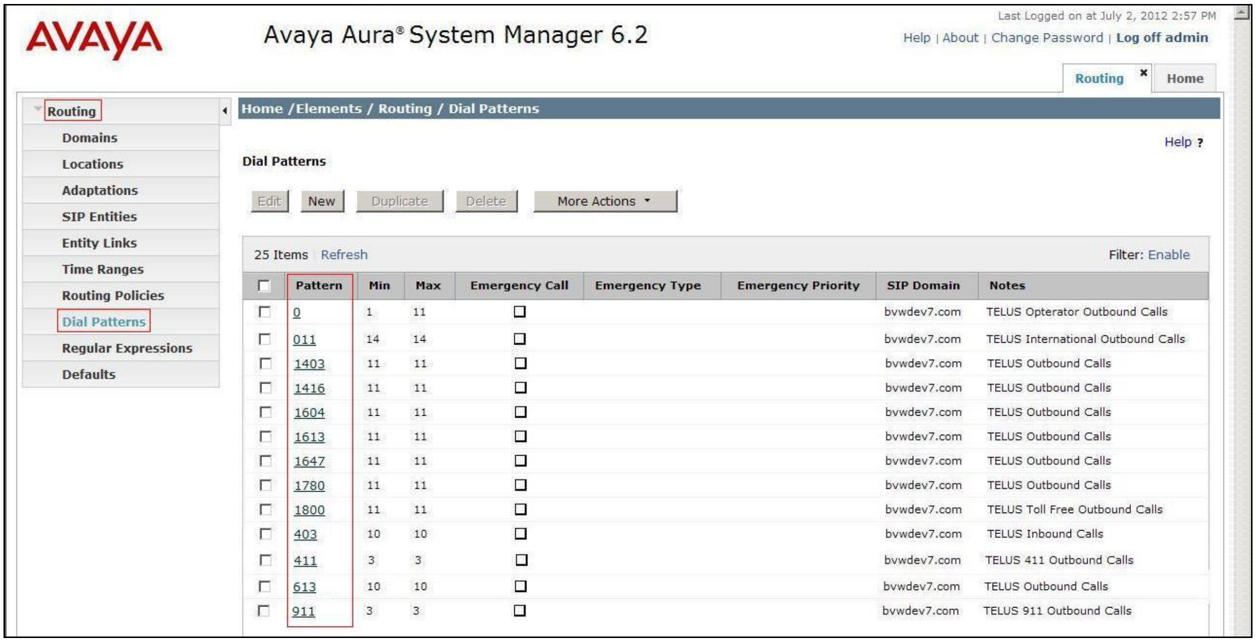
The following screen illustrates an example dial pattern used to verify inbound PSTN calls to the enterprise. When a user on the PSTN dials a number assigned to the TELUS Service, such as 403692xxxx, TELUS delivers the number to the enterprise, and the Avaya SBCE sends the call to Session Manager. Under **Originating Locations and Routing Policies**, the **Routing Policy Name To_Car3_ssg_carrier** is selected, which sends the call to Communication Server as described previously and **Routing Policy Destination** is set as **car3-ssg-carrier**.



The following screen illustrates an example dial pattern used to verify outbound calls from the enterprise to the PSTN. When a Communication Server 1000 user dials a PSTN number such as 1613-967-xxxx, Communication Server 1000 sends the call to Session Manager. Session Manager will match the dial pattern shown below and send the call to the Avaya SBCE via the **Routing Policy Name To_TELUS**. The **Routing Policy Destination** is set as **AvayaSBCE**.



The following screen illustrates an example **Dial Patterns** used to verify inbound and outbound calls between the enterprise and the PSTN.



7. Configure Avaya SBCE

This section describes the configuration of the Avaya SBCE necessary for interoperability with the Session Manager and TELUS systems.

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

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

7.1. Log in Avaya SBCE

Access the web interface by typing "**https://x.x.x.x**" (where x.x.x.x is the management IP of the Avaya SBCE).

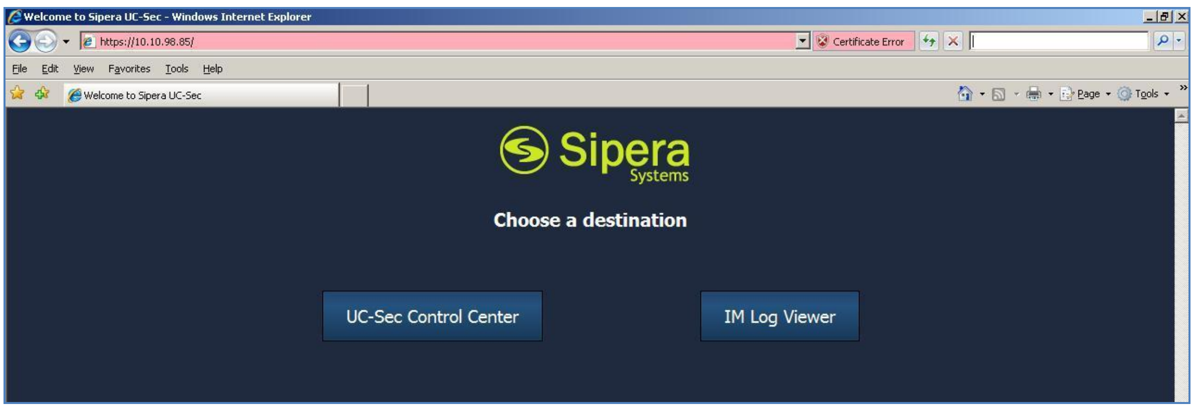


Figure 48: Avaya SBCE Web Interface

Select **UC-Sec Control Center** and enter the **Login ID** and **Password**.

Figure 49: Avaya SBCE Login

7.2. Global Profiles

When selected, Global Profiles allows for configuration of parameters across all UC-Sec appliances.

7.2.1. Configure Server Interworking - Avaya Side

Server Interworking allows you to configure and manage various SIP call server-specific capabilities such as call hold, 180 Handling, etc.
From the menu on the left-hand side, select **Global Profiles → Server Interworking → Add Profile**:

- Enter Profile name**: SM62**
- Check **Hold Support** as **RFC2543**
- Check **Diversion Header Support** as **Yes.**
- All other options on the General Tab can be left at default.

On the **Timers**, **URI Manipulation**, **Header Manipulation** and **Advanced** Tabs: All options can be left at default. Click **Finish** (not shown).
The Figure 50 is shown that Session Manager server interworking (named: SM62) was added.

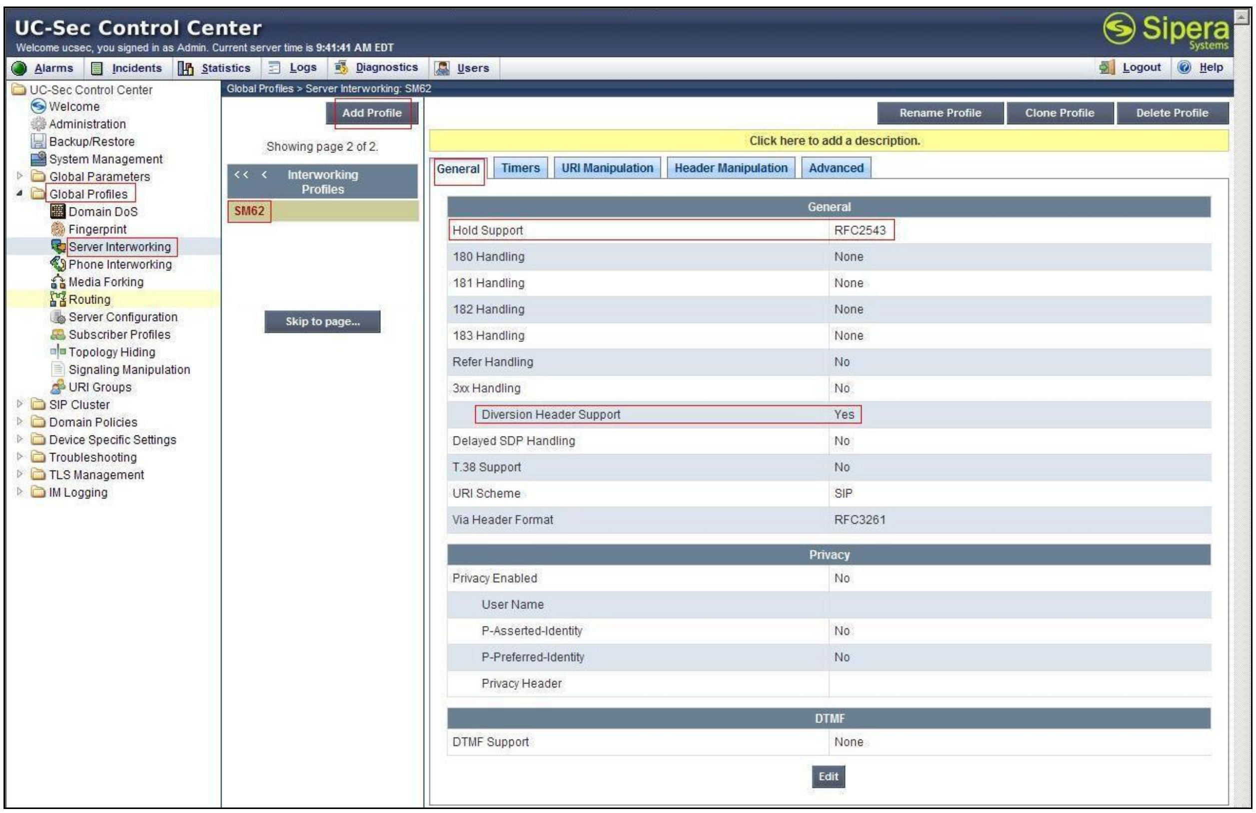


Figure 50: Server Interworking – Avaya Side

7.2.2. Configure Server Interworking – TELUS side

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

- Enter Profile name**: TELUS**
- Check **Hold Support** as **RFC2543**
- Check **Diversion Header Support** as **Yes**
- All other options on the General Tab can be left at default.

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

The Figure 51 is shown that TELUS server interworking (named: TELUS) was added.

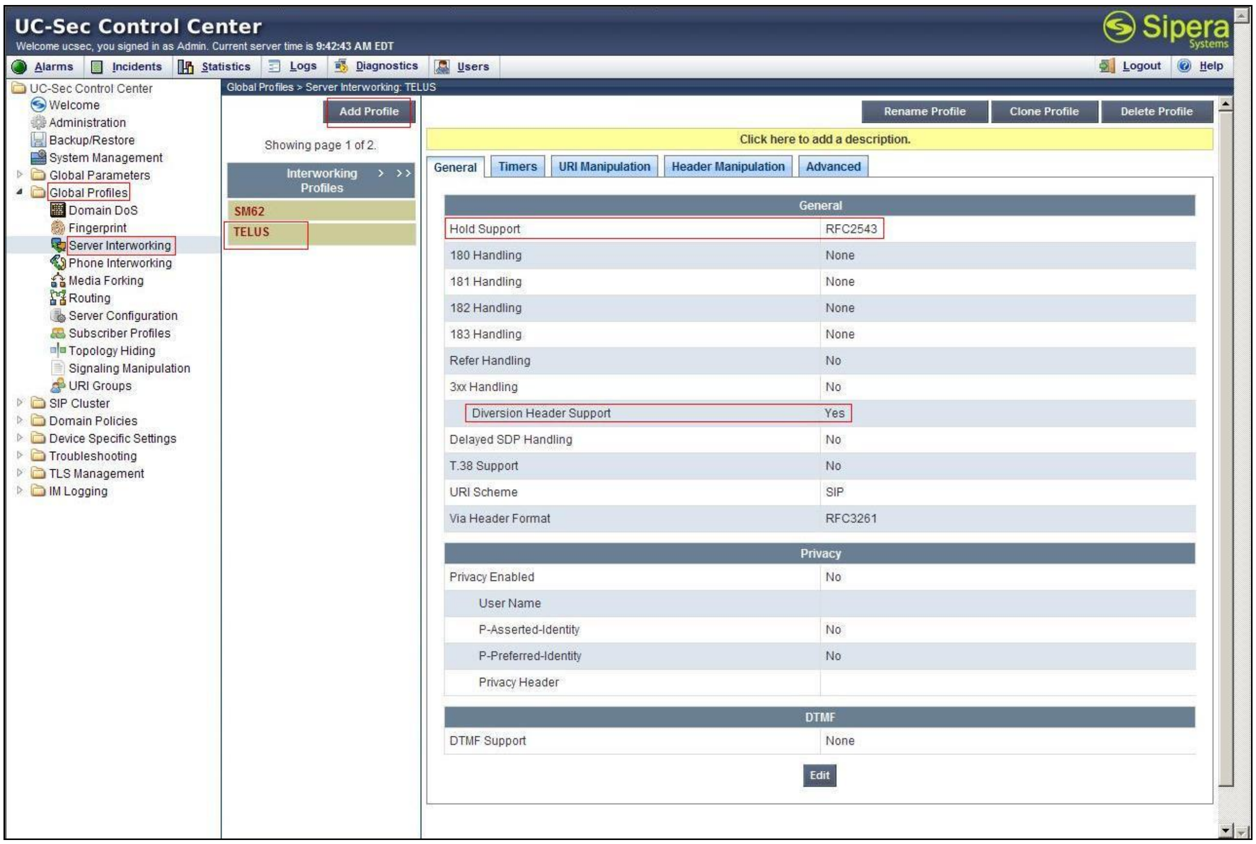


Figure 51: Server Interworking – TELUS Side

7.2.3. Configure Routing – Avaya side

Routing profiles define a specific set of packet routing criteria that are used in conjunction with other types of domain policies to identify a particular call flow and thereby ascertain which security features will be applied to those packets. Parameters defined by Routing Profiles include packet transport settings, name server addresses and resolution methods, next hop routing information, and packet transport types.

From the menu on the left-hand side, select **Global Profiles → Routing → Add Profile**
Enter Profile Name: **TELUS_To_SM62**

- **URI Group**: **TELUS_Group**
- **Next Hop Server 1: 10.10.97.198 (Session Manager IP address)**
- Check **Next Hop Priority**
- **Outgoing Transport: UDP**
- Click **Finish** (not shown).

Figure 52: Routing To Avaya

7.2.4. Configure Routing - TELUS side

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

From the menu on the left-hand side, select **Global Profiles → Routing →Add Profile**
Enter Profile Name: **SM62_To_TELUS**

- **URI Group**: **TELUS_Group**
- **Next Hop Server 1: 20.20.119.218** (IP Address provided by Customer**)**
- Check **Next Hop Priority**
- **Outgoing Transport** as **UDP**
- Click **Finish** (not shown).

Figure 53: Routing To TELUS

7.2.5. Configure Server – Session Manager

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

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

Enter profile name: **SM62**

On General tab (**Figure 54**):

- **Server Type:** Select **Call Server**
- **IP Address/FQDNs: 10.10.97.198 (Session Manager IP Address)**
- **Supported Transports**: **UDP**
- **UDP Port: 5060**

Figure 54: Session Manager Server Configuration 1

- On the **Advanced** tab (**Figure 55**), select **SM62** for **Interworking Profile**
- Click **Finish** (not shown).

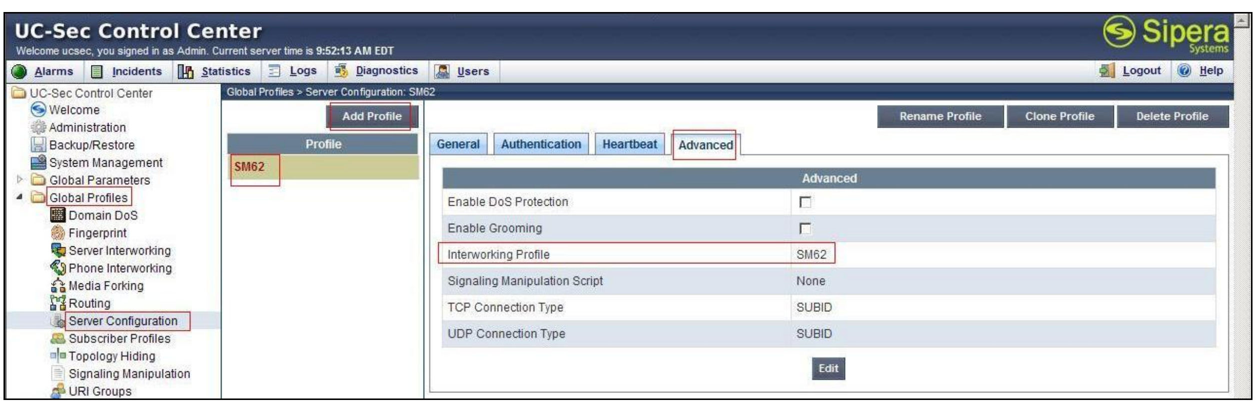


Figure 55: Session Manager Server Configuration 2

7.2.6. Configure Server – TELUS ACME packet SBC

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

Enter profile name: **TELUS**

On General tab (**Figure 56**):

- **Server Type:** Select **Trunk Server**
- **IP Address: 20.20.119.218** (TELUS Trunk Server)
- **Supported Transports**: **UDP**
- **UDP Port: 5060**

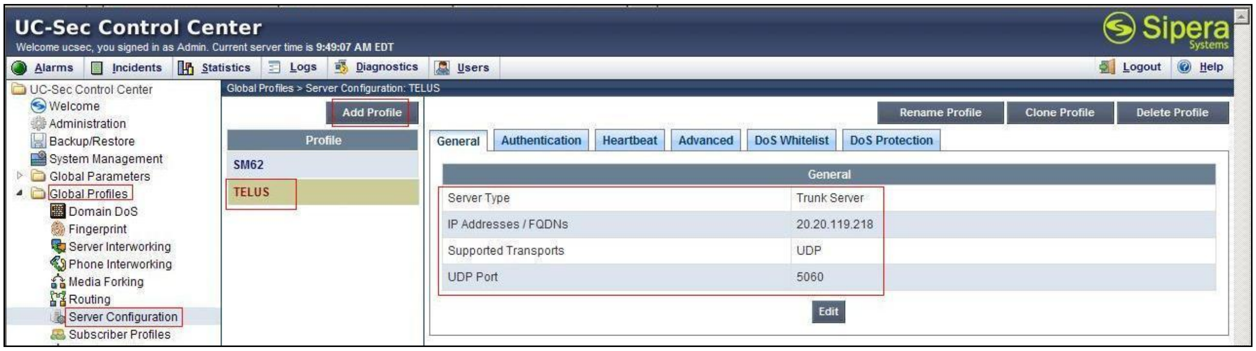


Figure 56: TELUS Server Configuration

On the **Advanced** Tab (**Figure 57**):

- Select **TELUS** for **Interworking Profile**
- Select **Signaling Manipulation Script**: **For TELUS**

Figure 57: TELUS Server Advanced Configuration

On the **Heartbeat** Tab (**Figure 58**):

- Check on **Enable Heartbeat**
- Select **Method** as **OPTIONS** (TELUS requires)
- **Frequency: 60 seconds**
- **From URI**: **ping@bvwdev7.com**
- **To URI**: **ping@20.20.119.218**

- Check **TCP Probe**, **TCP Probe Frequency**: **10 seconds**

Click **Finish** (not shown).

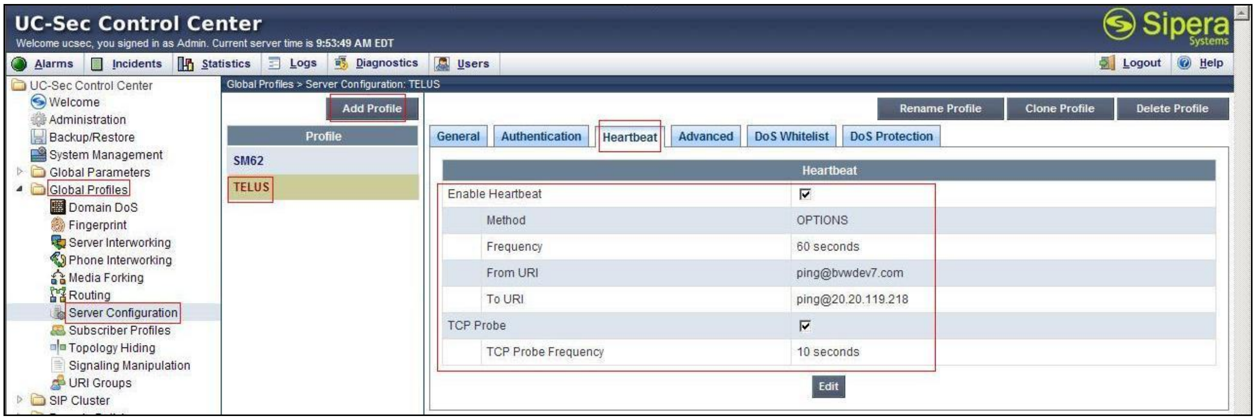


Figure 58: TELUS Server Heartbeat Configuration

7.2.7. Configure Topology Hiding – Avaya side

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

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

Select **Add Profile**, enter Profile Name: **TELUS_To_SM62**

- For the Header **To,**
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select**: Overwrite**
  - In the **Overwrite Value** column**: bvwdev7.com**
- For the Header **From,**
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select**: Overwrite**
  - In the **Overwrite Value** column**: bvwdev7.com**
- For the Header **Request-Line,**
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select**: Overwrite**
  - In the **Overwrite Value** column**: bvwdev7.com**

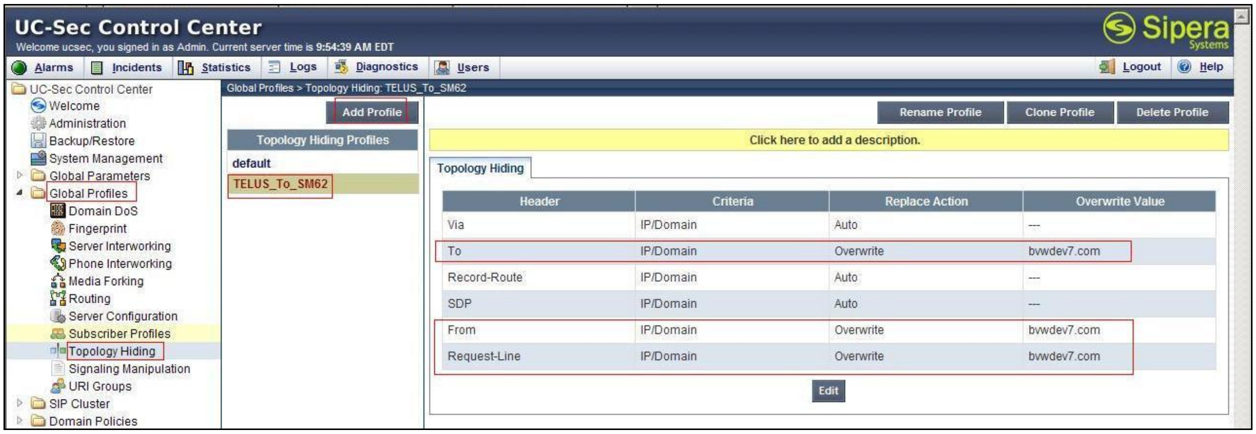


Figure 59: Topology Hiding Session Manager

7.2.8. Configure Topology Hiding – TELUS side

From the menu on the left-hand side, select **Global Profiles → Topology Hiding**
Select **Add Profile**, enter Profile Name: **SM62_To_TELUS**

- For the Header **To,**
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select**: Overwrite**
  - In the **Overwrite Value** column**: 20.20.119.218**
- For the Header **Request-Line,**
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select**: Overwrite**
  - In the **Overwrite Value** column**: 20.20.119.218**

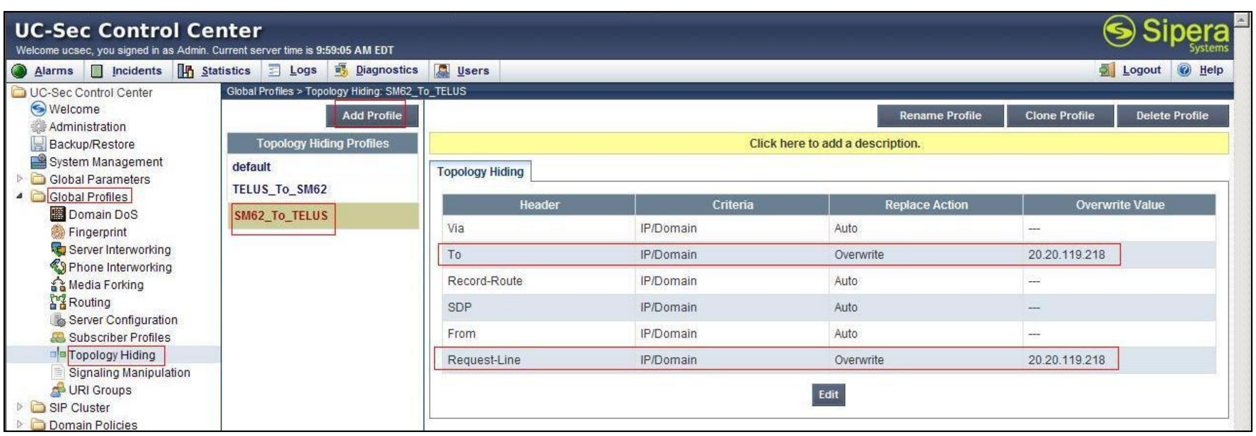


Figure 60: Topology Hiding TELUS

7.2.9. Configure Signaling Manipulation

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

From the menu on the left-hand side, select **Global Profiles → Signaling Manipulation →Add Script**.

Enter script Title: **For TELUS**

- Edit the script as **Figure 61**
  - To replace the Request Line <u>sip:domain</u> from the body in SIP message
  - To replace information of PAI field by information of Diversion Header field
  - To remove History Info
- Click Save (not shown).

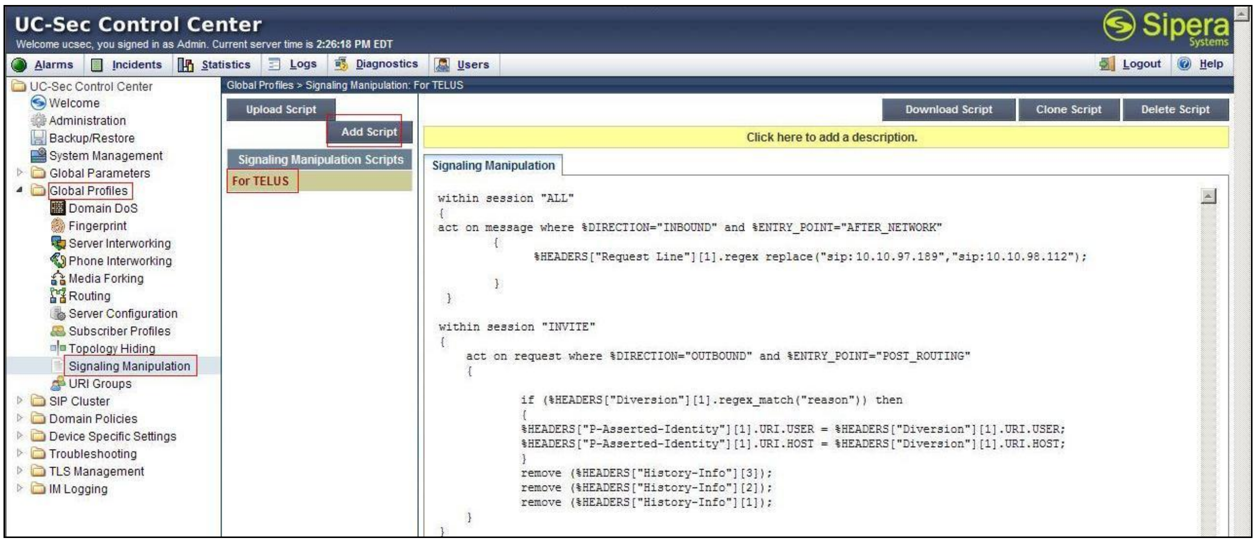


Figure 61: Signaling Manipulation

7.2.10. Configure URI Groups

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

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

- Select **Add Groups**, enter Group Name: **TELUS_Group**
- Edit the URI Type: Plain (not shown)
- Add URI: ***@*10.10.98.112**, ***@20.20.119.218** , ***@anonymous.invalid** , ***@bvwdev7.com**
- Click **Finish** (not shown).

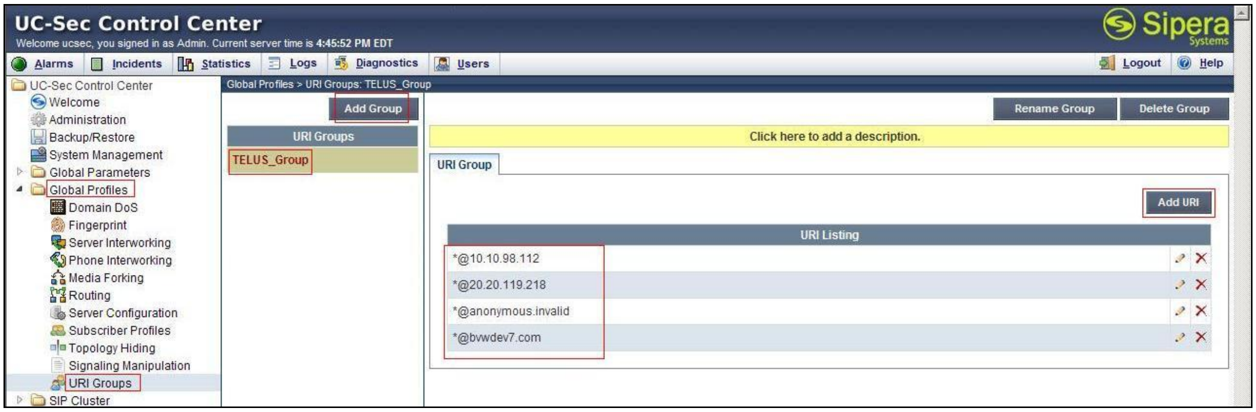


Figure 62: URI Group

7.3. Domain Policies

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

7.3.1. Create Application Rules

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

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

- Select the **default** Rule
- Select **Clone Rule** button
  - Name**: SM62_TELUS_AppR**
  - Click **Finish** (not shown).

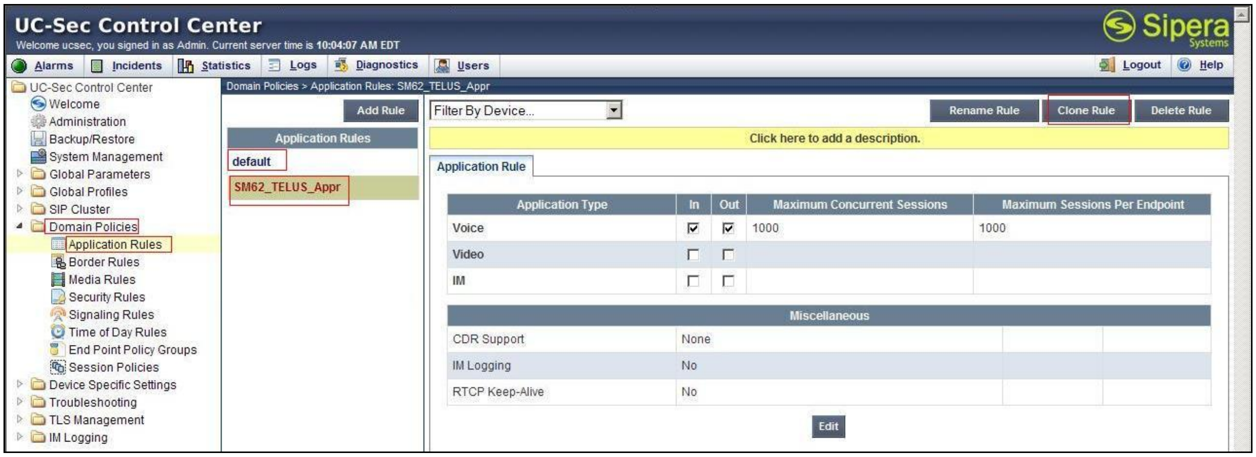


Figure 63: Session Manager Application Rule

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

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

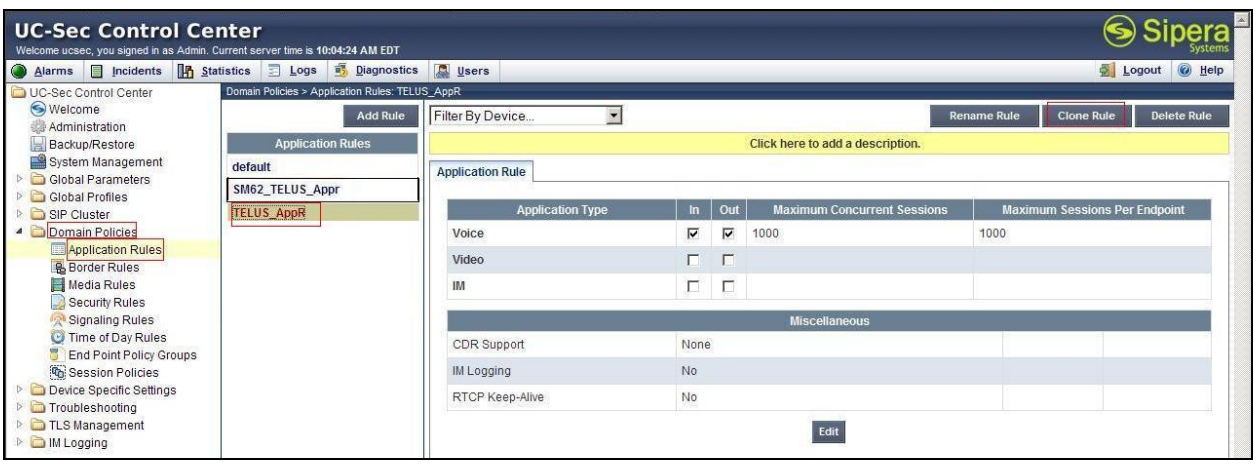


Figure 64: TELUS Application Rule

7.3.2. Create Border Rules

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

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

- Select the **default** Rule
- Select **Clone Rule** button
    - Enter Clone Name**: SM62_TELUS_BorderR**

- Click **Finish** (not shown)**.**

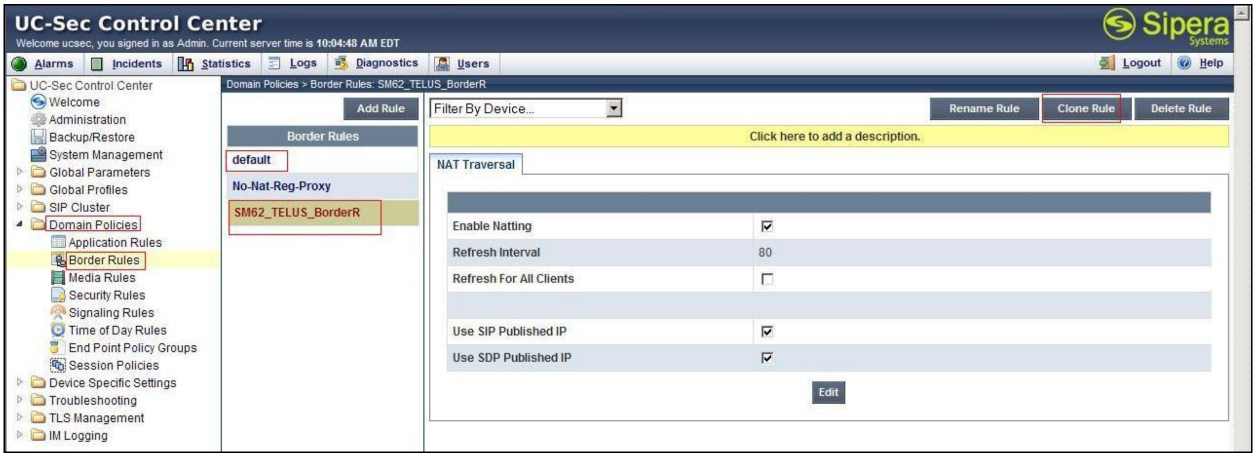


Figure 65: Session Manager Border Rule

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

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

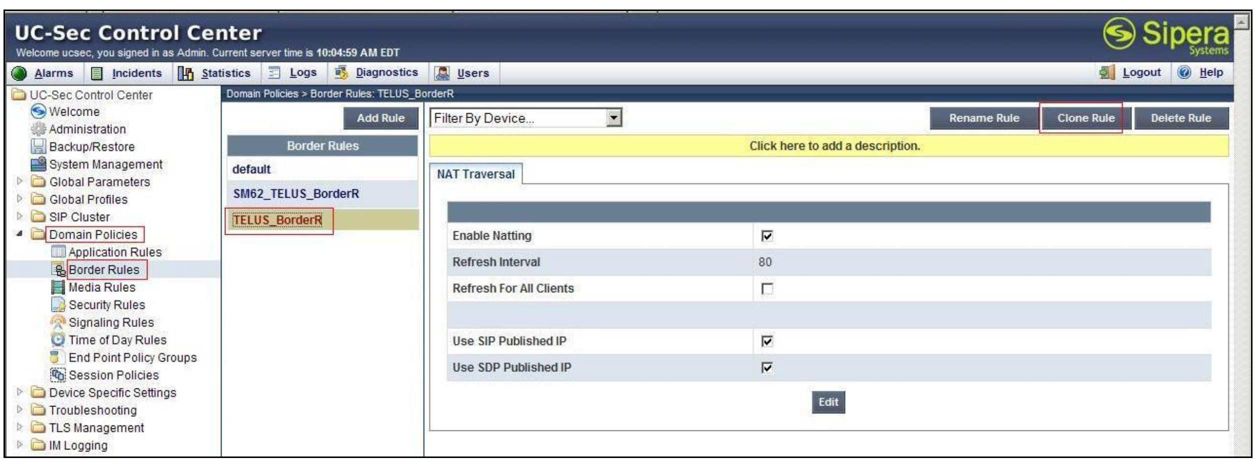


Figure 66: TELUS Border Rule

7.3.3. Create Media Rules

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

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

- Select the **default-low-med** Rule
- Select **Clone Rule** button
  - Enter Clone Name**: SM62_TELUS_MediaR**
  - Click **Finish** (not shown)**.**

Figure 67: Session Manager Media Rule

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

- Select the **default-low-med** Rule
- Select **Clone Rule** button
  - Enter Clone Name**: TELUS_MediaR**
  - Click **Finish** (not shown)

Figure 68: TELUS Media Rule

7.3.4. Create Security Rules

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

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

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

- Enter Clone Name**: SM62_TELUS_SecurityR**
- Click **Finish** (not shown)

Figure 69: Session Manager Security Rule

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

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

Figure 70: TELUS Security Rule

7.3.5. Create Signaling Rules

Signaling Rules allow you to define the action to be taken (*Allow*, *Block*, *Block with Response*, etc.) for each type of SIP-specific signaling request and response message. When SIP signaling packets are received by the UC-Sec, they are parsed and "patternmatched" against the particular signaling criteria defined by these rules. Packets matching the criteria defined by the Signaling Rules are tagged for further policy matching

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

- Select the **default** Rule
- Select **Clone Rule** button
  - Enter Clone Name**: SM62_TELUS_SigR**
  - Click **Finish** (not shown)**.**

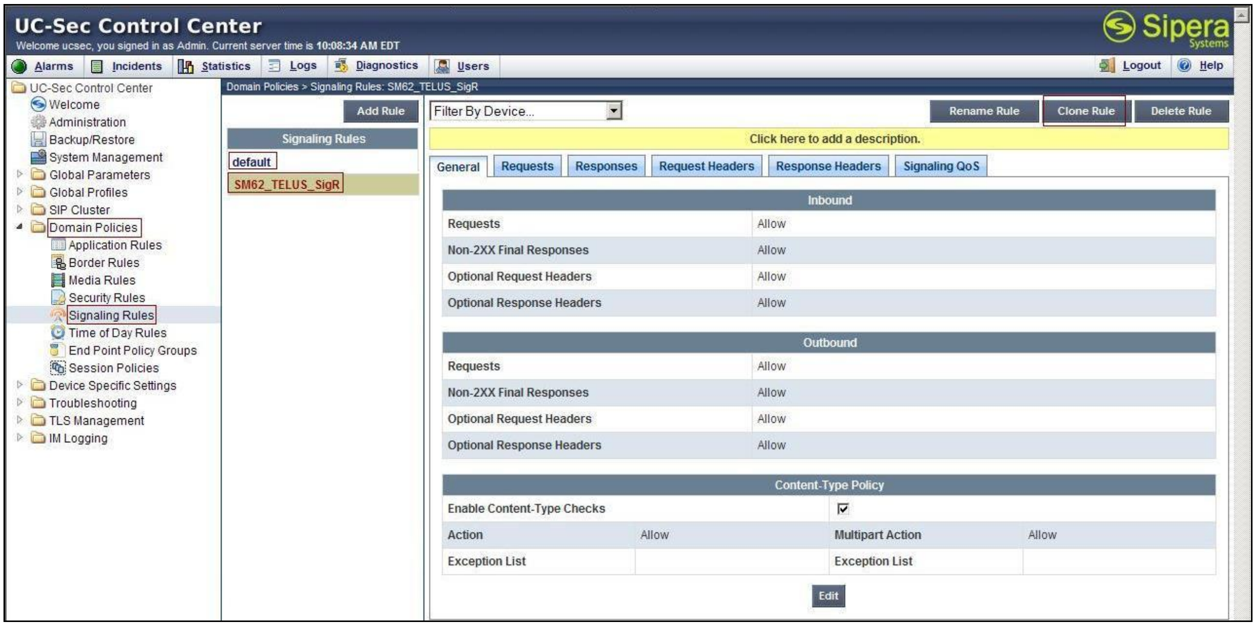


Figure 71: Session Manager Signaling Rule

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

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

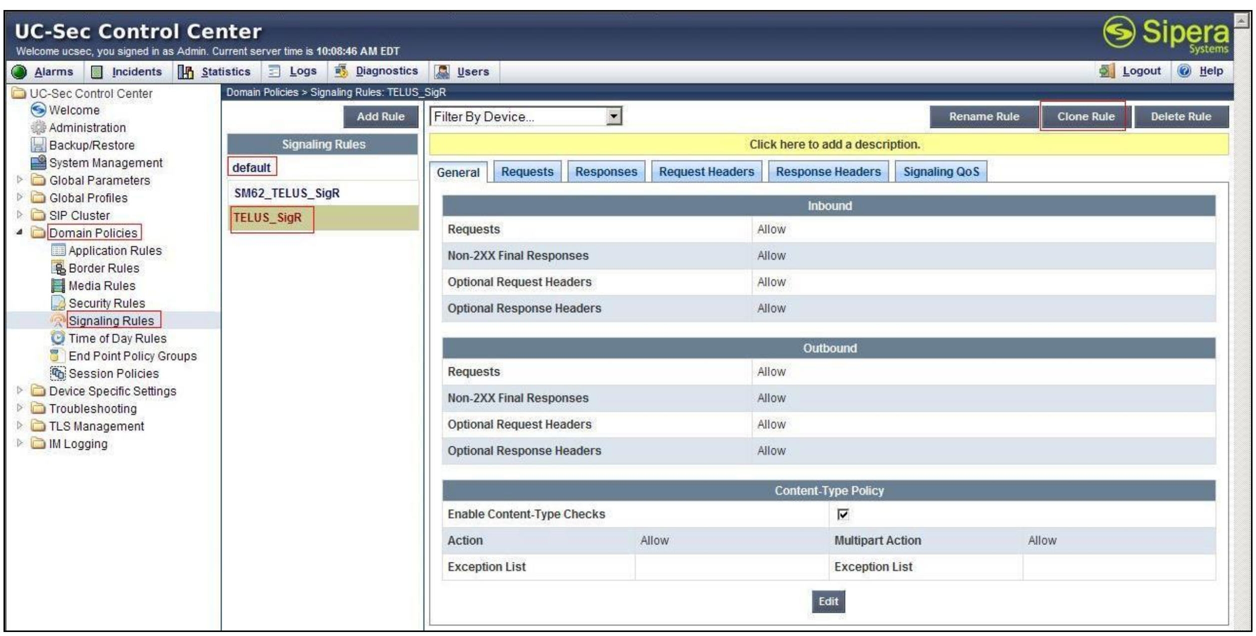


Figure 72: TELUS Signaling Rule 1

7.3.6. Create Time of Day Rules

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

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

- Select the **default** Rule
- Select **Clone Rule** button
  - Enter Clone Name**: SM62_TELUS_ToDR**
  - Click **Finish** (not shown)**.**

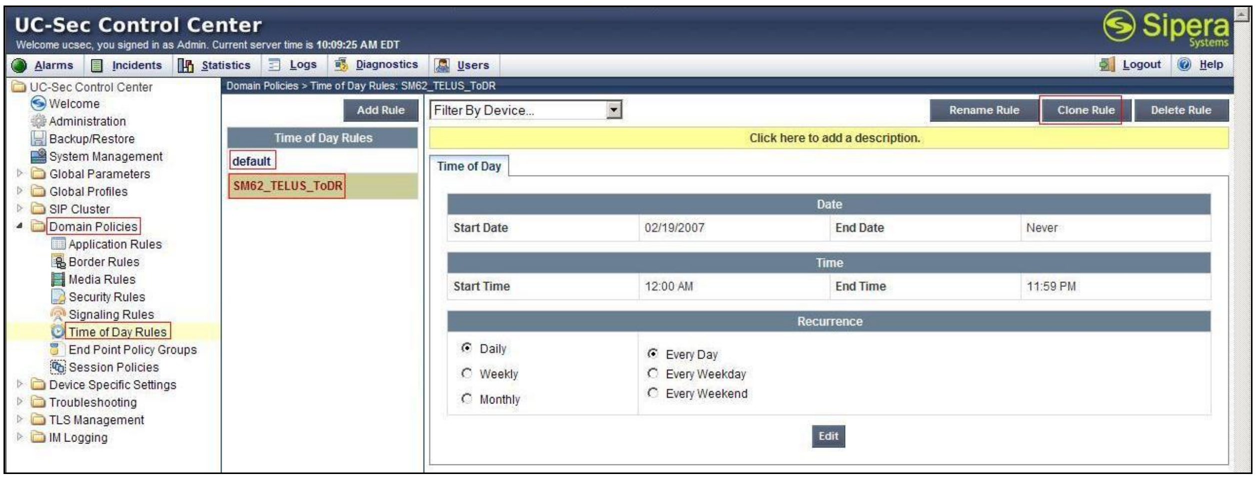


Figure 73: Session Manager Time of Day Rule

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

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

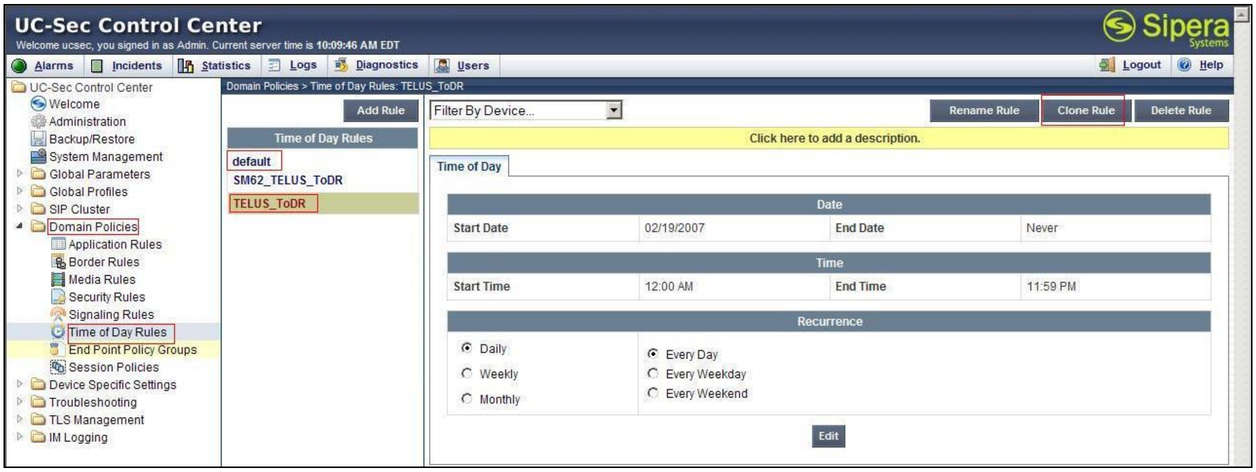


Figure 74: TELUS Time of Day Rule

7.3.7. Create Endpoint Policy Groups

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

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

- Select **Add Group**
- Enter **Group Name: SM62_TELUS_PolicyG**
  - **Application Rule: SM62_TELUS_AppR**
  - **Border Rule: SM62_TELUS _BorderR**
  - **Media Rule: SM62_TELUS _MediaR**
  - **Security Rule: SM62_TELUS _SecurityR**
  - **Signaling Rule: SM62_TELUS _SigR**
  - **Time of Day: SM62_TELUS _ToDR**

Select **Finish** (not shown)**.**

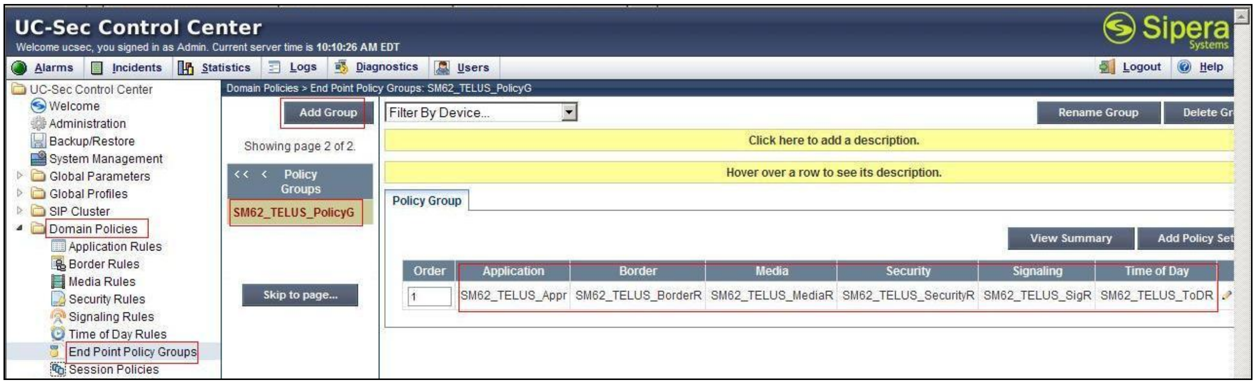


Figure 75: Session Manager End Point Policy Group

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

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

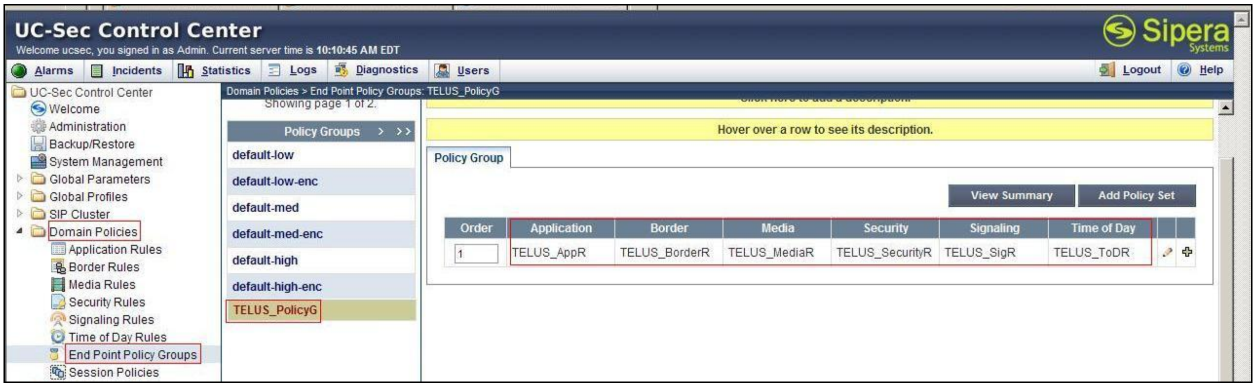


Figure 76: TELUS End Point Policy Group

7.4. Device Specific Settings

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

7.4.1. Manage Network Settings

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

- Enter the **IP Address** and **Gateway Address** for both the Inside and the Outside interfaces:
  - **IP Address for Inside interface: 10.10.97.189; Gateway: 10.10.97.129**
  - **IP Address for Outside interface: 10.10.98.112; Gateway: 10.10.98.97**
- Select the physical interface used in the Interface column:
  - **Inside Interface**: **A1**
  - **Outside Interface**: **B1**

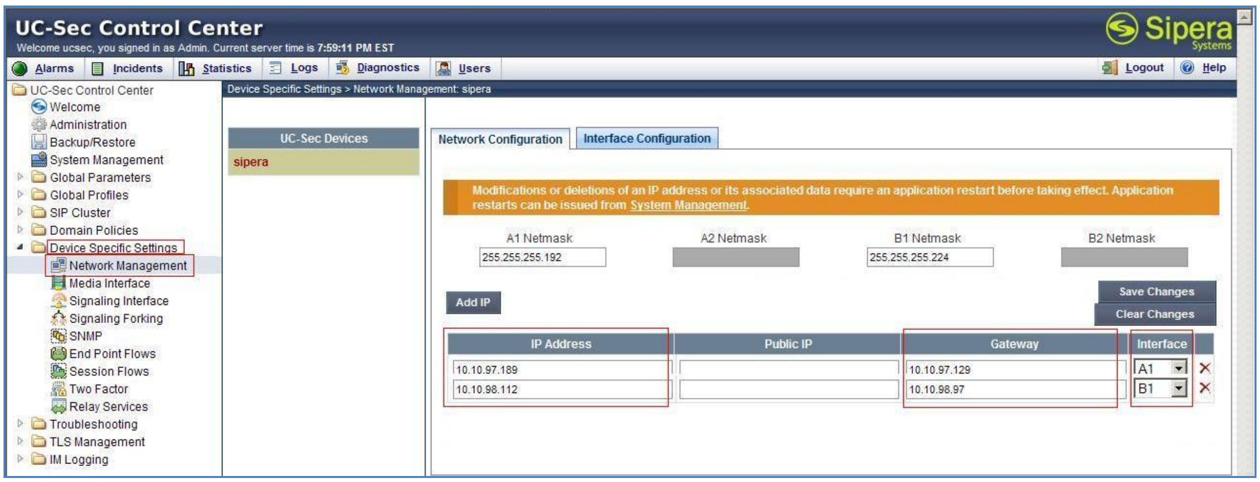


Figure 77: Network Management

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

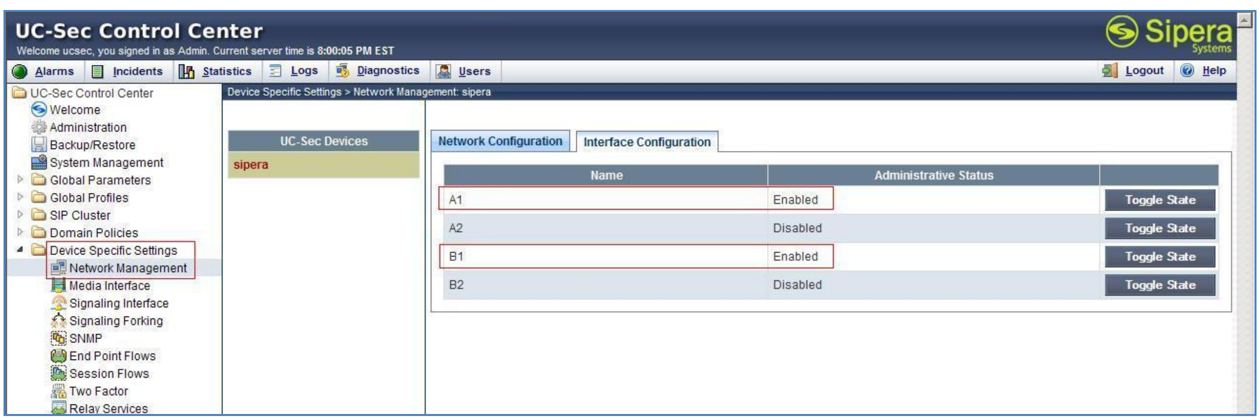


Figure 78: Network Interface Status

7.4.2. Create Media Interfaces

Media Interfaces (**Figure 79**) define the type of signaling on the ports. The default media port range on the Avaya can be used for both inside and outside ports

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

- Select **Add Media Interface**
  - **Name: InsideMedia**
  - **Media IP: 10.10.97.189** (Internal Address toward Session Manager)
  - **Port Range: 35000 - 40000**
  - Click **Finish** (not shown)
- Select **Add Media Interface**
  - **Name: OutsideMedia_SBCE**
  - **Media IP: 10.10.98.112** (External Internet Address toward TELUS trunk)
  - **Port Range: 35000 - 40000**
  - Click **Finish** (not shown).

Figure 79: Media Interface

7.4.3. Create Signaling Interfaces

Signaling Interfaces (**Figure 80**) define the type of signaling on the ports

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

- Select **Add Signaling Interface**
  - **Name: InsideSIP**
  - **Media IP: 10.10.97.189 (Internal Address toward Session Manager)**
  - **UDP Port: 5060**
  - Click **Finish** (not shown)

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

- Select **Add Signaling Interface**
  - **Name: OutsideSIP_SBCE**
  - **Media IP: 10.10.98.112 (External Internet Address toward TELUS trunk)**
  - **UDP Port: 5060**
  - Click **Finish** (not shown).

Figure 80: Signaling Interface

7.4.4. Configuration Server Flows

Server Flows (**Figure 81**) allow to categorize trunk-side signaling and apply a policy.

7.4.4.1 Create End Point Flows - Session Manager

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

- Select the **Server Flows** Tab
- Select **Add Flow,** enter **Flow Name: TELUS**
  - **Server Configuration**: **SM62**
  - **URI Group: TELUS_Group**
  - **Transport: ***
  - **Remote Subnet: ***
  - **Received Interface**: **OutsideSIP_SBCE**
  - **Signaling Interface: InsideSIP**
  - **Media Interface**: **InsideMedia**
  - **End Point Policy Group: SM62_TELUS_PolicyG**
  - **Routing Profile: SM62_To_TELUS**
  - **Topology Hiding Profile: TELUS_To_SM62**
  - **File Transfer Profile: None**
  - Click **Finish** (not shown).

7.4.4.2 Create End Point Flows – TELUS

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

- Select the **Server Flows** Tab
- Select **Add Flow**, enter **Flow Name: TELUS**
  - **Server Configuration**: **TELUS**
  - **URI Group: TELUS_Group**
  - **Transport: ***
  - **Remote Subnet: ***
  - **Received Interface**: **InsideSIP**
  - **Signaling Interface: OutsideSIP_SBCE**
  - **Media Interface**: **OutsideMedia_SBCE**
  - **End Point Policy Group: TELUS_PolicyG**
  - **Routing Profile: TELUS_To_SM62**

- **Topology Hiding Profile: SM62_To_TELUS**
- **File Transfer Profile: None**
- Click **Finish** (not shown)

Figure 81: End Point Flows

8. Verification Steps

The following steps may be used to verify the configuration

8.1. General

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

8.2. Verification of an Active Call on Call Server

Active Call Trace (LD 80)

The following is an example of one of the commands available on the Communication Server 1000 to trace the DN for which the call is in progress or idle. The call scenario involved PSTN phone number 6139675205 calling 4036929464.

- Login on to Signaling Server 10.10.97.177 with admin account and password.
- Issue a command "cslogin" to login on to the Call Server.
- Log in to the Overlay command prompt, issue the command **LD 80** and then **trace 0 9464**.
- After the call is released, issue command **trac 0 9464** again to see if the DN is released back to idle state.

Below is the actual output of the Call Server Command Line mode when the 9464 is in call state:

```
>ld 80

.trac 0 9464

ACTIVE  VTN 096 0 00 02
```

```
ORIG   VTN 100 0 00 00   VTRK IPTI  RMBR  100 1 INCOMING VOIP GW CALL
  FAR-END SIP SIGNALLING IP: 20.20.119.218
  FAR-END MEDIA ENDPOINT IP: 10.10.97.242  PORT: 24574
  FAR-END VendorID: Not available
TERM   VTN 096 0 00 02  KEY 0  SCR MARP  CUST 0  DN 9464  TYPE 2002P2
  SIGNALLING ENCRYPTION: INSEC
  MEDIA ENDPOINT IP: 10.10.98.3  PORT: 5200
MEDIA PROFILE: CODEC G.711 MU-LAW  PAYLOAD 20 ms  VAD OFF
RFC2833:  RXPT  101   TXPT  101   DIAL DN 9464
MAIN_PM  ESTD
TALKSLOT  ORIG  20   TERM  25
EES_DATA:
NONE
QUEU  NONE
CALL ID 501 76


----  ISDN ISL CALL (ORIG) ----
CALL REF # =  484
BEARER CAP =  VOICE
HLC =
CALL STATE =  10     ACTIVE
CALLING NO =  16139675205  NUM_PLAN:UNKNOWN     TON:UNKNOWN   ESN:UNKNOWN
CALLED NO  =  4036929464  NUM_PLAN:UNKNOWN     TON:UNKNOWN   ESN:UNKNOWN
```

And this is the example after the call on 9464 is finished.

```
.trac 0 9464
IDLE VTN 96 0 00 02   MARP
```

SIP Trunk monitoring (LD 32)

Place a call inbound from PSTN (6139675205) to an internal device (4036929464). Then check the SIP trunk status by using LD 32, one trunk is BUSY

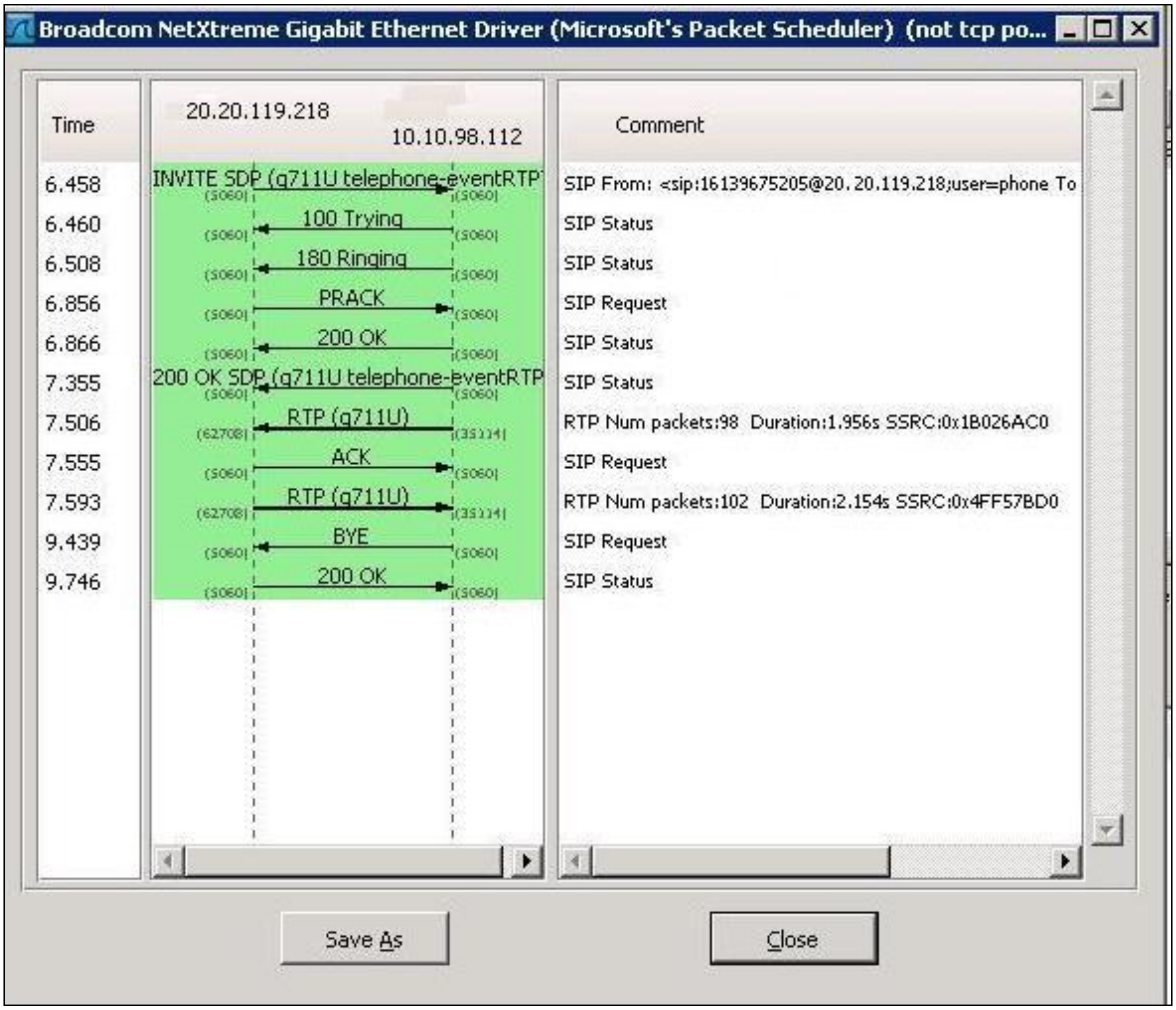
```
>ld 32
NPR000
.stat 100 0
091 UNIT(S) IDLE
001 UNIT(S) BUSY
000 UNIT(S) DSBL
000 UNIT(S) MBSY
```

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

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

8.3. Protocol Trace

Below is a wireshark trace of the same call scenario described in **Section 8.2**. Note that only detail of the INVITE message is being shown here.



```
⊟ Session Initiation Protocol
  ■ Request-Line: INVITE sip:4036929464@10.10.98.112:5060 SIP/2.0
      Method: INVITE
    ⊞ Request-URI: sip:4036929464@10.10.98.112:5060
      [Resent Packet: False]
  ⊟ Message Header
      Via: SIP/2.0/UDP 20.20.119.218:5060;branch=z9hG4bK610ek300e0jg2kc1e3c0.1
    ⊟ To: <sip:4036929464@10.10.98.112>
      ⊟ SIP to address: sip:4036929464@10.10.98.112
          SIP to address User Part: 4036929464
          SIP to address Host Part: 10.10.98.112
    ⊟ From: <sip:16139675205@20.20.119.218;user=phone>;tag=snl_0010398373_NSN_CLIENT
      ⊞ SIP from address: sip:16139675205@20.20.119.218;user=phone
        SIP tag: snl_0010398373_NSN_CLIENT
      Call-ID: NSNSIP-e88b19ac-e98b19ac-1-11-1341866922-470108-1342337030
    ⊞ CSeq: 1235 INVITE
    ⊞ Contact: <sip:16139675205@20.20.119.218:5060;transport=udp>
      Supported: 100rel
      Supported: timer
      Accept-Language: en;q=0.0
      Allow: REGISTER, INVITE, ACK, BYE, CANCEL, NOTIFY, REFER, INFO, PRACK
      Session-Expires: 1800;refresher=uac
      Min-SE: 1800
      Date: Mon, 09 Jul 2012 20:48:42 GMT
      Max-Forwards: 68
      Content-Type: application/sdp
      Content-Length: 209
  ⊟ Message Body
    ⊟ Session Description Protocol
        Session Description Protocol Version (v): 0
      ⊞ Owner/Creator, Session Id (o): PVG 1341866672580 1341866672580 IN IP4 20.20.119.218
        Session Name (s): -
        Phone Number (p): +1 6135555555
      ⊞ Connection Information (c): IN IP4 20.20.119.218
      ⊞ Time Description, active time (t): 0 0
      ⊞ Media Description, name and address (m): audio 62708 RTP/AVP 0 101
```

9. Conclusion

All of the test cases have been executed. Despite the number of observations seen during testing as noted in **Section 2.2**, the test result met the objectives outlined in **Section 2.1**. The TELUS system is considered **compliant** with Communication Server 1000 Release 7.5, Avaya Aura® Session Manager Release 6.2 and Avaya Session Border Controller for Enterprise Release 4.0.5 Q09

10. Additional References

Product services for Avaya SBCE may be found at:
http://www.sipera.com/products-services/esbc

Product documentation for Avaya, including the following, is available at:
http://support.avaya.com/

[1] *Network Routing Service Fundamentals, Avaya Communication Server 1000, Release 7.5, Document Number NN43001-130, Revision 03.10, September 2011.*

[2] *IP Peer Networking Installation and Commissioning, Avaya Communication Server 1000, Release 7.5, Document Number NN43001-313, Revision: 05.09, October 2011*

[3] *Communication Server 1000E Overview, Avaya Communication Server 1000, Release 7.5, Document Number NN43041-110, Revision: 05.05, October 2011*

[4] *Unified Communications Management Common Services Fundamentals, Avaya Communication Server 1000, Release 7.5, Document Number NN43001-116, Revision 05.17, January 2012*

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

[6] *Product Compatibility Reference, Avaya Communication Server 1000, Release 7.5, Document Number NN43001-256, Revision 05.03, December 2011*

[7] *Administering Avaya Aura® Session Manager, Doc ID 03-603324, Release 6.2, July* 2012

[8] *Administering Avaya Aura® System Manager, Release 6.2, July 2012*

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.