



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

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# **Application Notes for Configuring MTS Allstream SIP Trunk Service with Avaya Communication Server 1000 Release 7.6, Avaya Aura<sup>®</sup> Session Manager Release 6.3 and Avaya Session Border Controller for Enterprise Release 6.3 – Issue 1.0**

## **Abstract**

These Application Notes describe the procedure for configuration of the MTS Allstream SIP Trunk Service with Avaya Communication Server 1000 Release 7.6, Avaya Aura<sup>®</sup> Session Manager Release 6.3 and Avaya Session Border Controller for Enterprise Release 6.3.

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

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

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

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# 1. Introduction

These Application Notes illustrate a sample configuration using Avaya Communication Server 1000 (CS1000) Release 7.6, Avaya Aura<sup>®</sup> Session Manager Release 6.3 and Avaya Session Border Controller for Enterprise (Avaya SBCE) Release 6.3 with MTS Allstream SIP Trunk Service. MTS Allstream SIP Trunk Service provides PSTN access via SIP Trunks between the enterprise and the MTS Allstream SIP Trunk Service's network as an alternative to legacy analog or digital trunks.

## 2. General Test Approach and Test Results

CS1000 was connected to Avaya SBCE via Session Manager by using SIP Trunks. Avaya SBCE was connected to MTS Allstream SIP Trunk Service's network via SIP trunks. Various call types were made from CS1000 to MTS Allstream SIP Trunk Service and vice versa to verify interoperability.

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

### 2.1. Interoperability Compliance Testing

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

- General call processing between CS1000 and MTS Allstream SIP Trunk Service, including the following:
  - Codec/ptime (G.711 a-law/20ms, G.711 mu-law/20ms, and G.729/20ms), no Voice Activity Detection (VAD).
  - Calling Line Identification Display (CLID).
  - Ring-back tone.
  - Speech (audio) path.
- Incoming PSTN calls to various phone types including H.323, SIP, digital, and analog telephones at the enterprise. All inbound PSTN calls were routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN calls from various phone types including H.323, SIP, digital, and analog telephones at the enterprise. All outbound PSTN calls were routed from the enterprise across the SIP trunk to the service provider.
- Inbound and outbound PSTN calls to/from Avaya IP Softphone 2050.
- Dialing plan support: local, long distance, international, outbound toll-free, 911 Emergency, 411 Service, and Operator Assisted Call.
- Call redirection verification: all supported methods (blind transfer, consultative transfer, call forward, and conference). Call redirection was performed from both ends. Note: MTS Allstream SIP Trunk Service supports Diversion Header for off-net call forwarding.

- Response to SIP OPTIONS queries.
- Response to incomplete call attempts and trunk errors.
- Fax T.38 and G.711 pass-through.
- Inbound and outbound long hold time call stability.
- Privacy requests (i.e., caller anonymity) and Caller ID restriction for inbound and outbound calls.
- DTMF (RFC2833) in inbound and outbound calls.
- SIP Transport UDP, port 5060.
- Voicemail navigation for inbound and outbound calls.
- CS1000 Mobile-X feature.
- Early Media Transmission.

The following is item not tested:

- Inbound toll-free is supported but was not tested as part of the compliance test.

The following assumptions were made for the compliance tested configuration:

- CS1000 R7.6 software with latest patches.
- MTS Allstream SIP Trunk Service provides support to setup, configure and troubleshoot on carrier switch during testing execution.

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

- Calls were checked for the correct call progress tones and cadences.
- During the ringing state, the ring back tone and destination ringing were checked.
- Calls were checked in both hands-free and handset mode due to internal Avaya requirement.
- Calls were checked for speech path in both directions using spoken words to ensure clarity of speech.
- 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.
- The speech path and messaging system were observed for timely and quality End-to-End tone audio path generation and application responses.
- The call server maintenance terminal window was open during the test execution for the monitoring of BUG(s), ERROR and AUD messages (See **Section 5.1.2**).
- Speech path was checked before and after calls were put on/off hold from each end.
- Calls were checked to ensure that all resources such as Virtual trunks, TDM trunks, Sets and VGWs (Voice Gateways) were released when calls were ended (See SIP Trunk monitoring in **Section 9.2**).

## 2.2. Test Results

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

- **The Calling Line Identification Display (CLID) was not available after hold/resume** – If the CS1000 phone holds/resumes an outbound call, the dialed digits were no longer displayed. This is a CS1000 known issue.
- **There is no ring-back tone after the off-net blind transfer is completed - PSTN1** phone calls an Avaya Communication Server 1000 phone, the user could not press the transfer button on the Avaya Communication Server 1000 phone to complete a blind transfer to PSTN2. In this particular scenario, SIP UPDATE support was required on the Communication Server 1000 for blind transfer, but for some reason, the SIP UPDATE on the PSTN-to-SIP gateway that MTS Allstream service used for this interoperability testing was not supported. In order to resolve this, plug-in 501 was enabled on the Communication Server 1000 to allow blind transfer to work without the UPDATE method (On CS1000 Element Manager, select **System** → **Software** → **Plug-ins** and then click on number **501** to enable plug-in 501). After the user was able to press the transfer button on the Avaya Communication Server 1000 to complete blind transfer, the PSTN1 phone could not hear ring-back-tone from the PSTN2. This is a CS1000 limitation.
- **There is no ring-back tone in Mobile-X phone to PSTN** – Mobile-X phone dials MSA (Mobile Service Access) number, and then dials any PSTN phone number. Mobile-X phone could not hear ring-back-tone from the PSTN. This is CS1000 limitation.

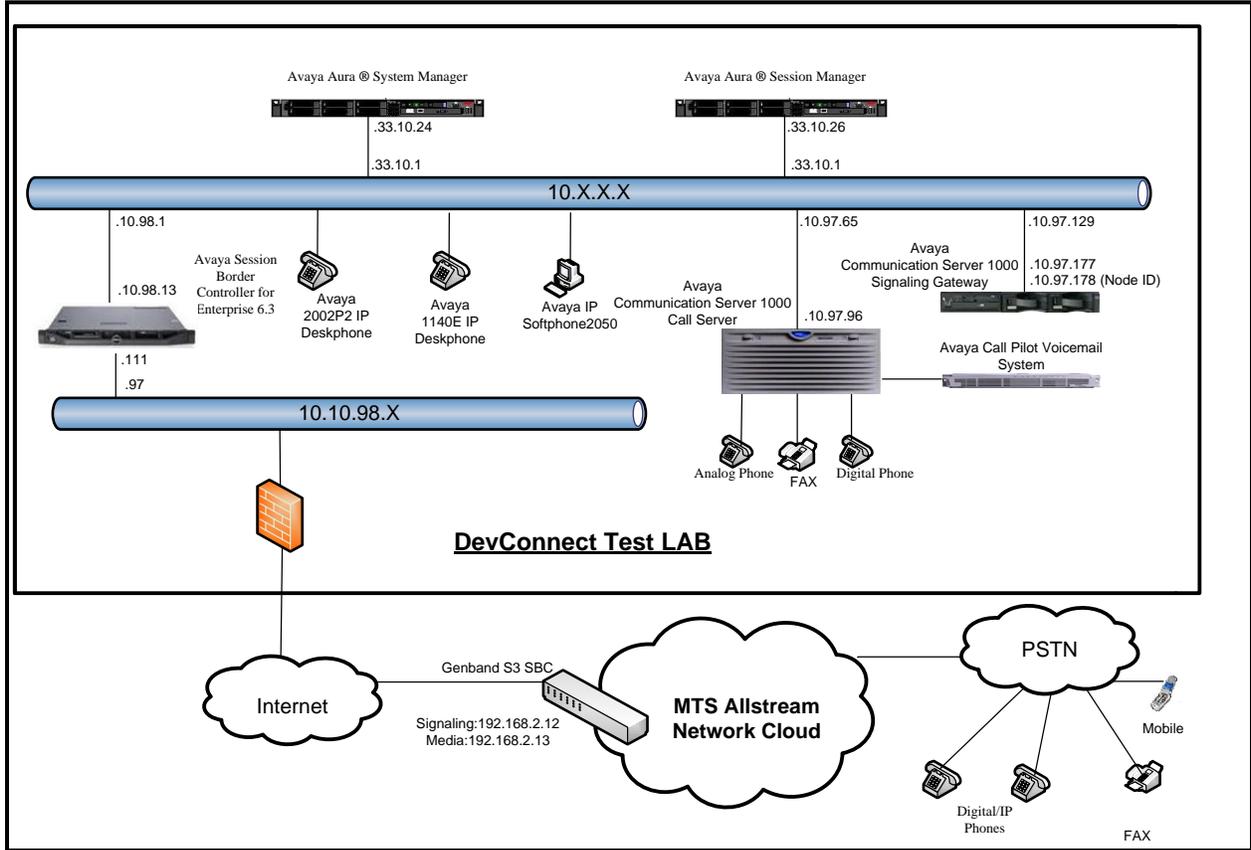
## 2.3. Support

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

For technical support on the MTS Allstream SIP Trunk Service, please contact customer service at 855-299-7050 or visit: <http://www.allstream.com/support>.

### 3. Reference Configuration

**Figure 1** illustrates the test configuration used during the compliance test between CS1000 and MTS Allstream SIP Trunk Service. For confidentiality and privacy purposes, actual public IP addresses used in this testing have been masked and replaced with fictitious IP addresses throughout the document.



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

## 4. Equipment and Software Validated

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

### Avaya systems:

Equipment/Software	Release/Version
Avaya Communication Server 1000 (CPPM)	Call Server: 765 P + Signaling Server: 7.65.16 GA SIP Line Server: 7.65.16 GA
Avaya Call Pilot C201i	Call Pilot Voice Mail Manager: 05.00.41.143
Avaya S8800 Server	Avaya Aura <sup>®</sup> Session Manager R6.3.7 – 6.3.7.0.637008
Avaya S8800 Server	Avaya Aura <sup>®</sup> System Manager R6.3.10 Build No. - 6.3.0.8.5682-6.3.8.4514 (Software Update Revision No: 6.3.10.7.2656)
Avaya Session Border Controller for Enterprise	6.3.000-19-4338
Avaya 2002 P2 IP Deskphone Avaya 1140E IP Deskphone	0604DCO 04.03.12.00 (SIP firmware)
Avaya 3904 Digital Phone	N/A
Avaya IP Softphone 2050	4.04.0067
Analog Symphony 2000	N/A
HP Office jet 4500 Fax	N/A

### MTS Allstream SIP Trunk Service systems:

System	Software
Genband S3 SBC	7.1.15.2
Genband CS2K Hybrid SoftSwitch	CVM17

Additional patch lineup for the configuration is listed as follows:

**Call Server:** 7.65 P+ GA plus latest DEPLIST – CPM\_7.6\_6.zip (X2107.65P)

**Signaling Server:** 7.65.16 GA plus latest DEPLIST – SP\_7.6\_6.ntl (7.65.16.00)

CS1000 Signaling Server patch list:

```
[admin@car3-cores ~]$ pstat
```

```
Product Release: 7.65.16.00
```

```
In system patches: 5
```

```
PATCH# NAME IN_SERVICE DATE SPECINS TYPE RPM
38 p31484_1 Yes 20/02/14 NO FRU cs1000-shared-general-7.65.16-00.i386
47 p33125_1 Yes 23/12/14 NO FRU cs1000-OS-1.00.00.00-00.noarch
48 p33274_1 Yes 23/12/14 YES FRU initscripts-8.45.25-1.el5.i386
49 p33331_1 Yes 23/12/14 YES FRU cs1000-OS-1.00.00.00-00.noarch
50 p33384_1 Yes 23/12/14 NO FRU cs1000-OS-1.00.00.00-00.noarch
```

In System service updates: 31

PATCH#	IN_SERVICE	DATE	SPECINS	REMOVABLE	NAME
0	Yes	23/12/14	YES	YES	cs1000-linuxbase-7.65.16.23-3.i386.000
1	Yes	23/12/14	NO	YES	cs1000-Jboss-Quantum-7.65.16.23-3.i386.000
2	Yes	23/12/14	YES	YES	cs1000-patchWeb-7.65.16.22-4.i386.000
3	Yes	23/12/14	YES	YES	cs1000-dmWeb-7.65.16.23-1.i386.000
4	Yes	23/12/14	YES	YES	cs1000-csoneksvrmgr-7.65.16.22-5.i386.000
5	Yes	23/12/14	YES	YES	cs1000-baseWeb-7.65.16.22-4.i386.000
6	Yes	23/12/14	YES	YES	cs1000-oam-logging-7.65.16.22-4.i386.000
7	Yes	23/12/14	YES	YES	cs1000-csv-7.65.16.22-2.i386.000
8	Yes	23/12/14	YES	YES	cs1000-mscTone-7.65.16.22-2.i386.000
9	Yes	23/12/14	YES	YES	cs1000-mscMusc-7.65.16.22-4.i386.000
10	Yes	23/12/14	YES	YES	cs1000-mscConf-7.65.16.22-2.i386.000
11	Yes	23/12/14	YES	YES	cs1000-mscAnnc-7.65.16.22-2.i386.000
12	Yes	23/12/14	YES	YES	cs1000-mscAttn-7.65.16.22-2.i386.000
13	Yes	23/12/14	NO	YES	cs1000-gk-7.65.16.22-1.i386.000
14	Yes	23/12/14	YES	YES	cs1000-shared-pbx-7.65.16.22-3.i386.000
15	Yes	20/02/14	NO	YES	cs1000-pd-7.65.16.21-00.i386.000
16	Yes	20/02/14	NO	YES	cs1000-shared-carrdtct-7.65.16.21-01.i386.000
17	Yes	20/02/14	NO	YES	cs1000-shared-tpselect-7.65.16.21-01.i386.000
18	Yes	20/02/14	NO	yes	cs1000-dbcom-7.65.16.21-00.i386.000
26	Yes	20/02/14	NO	YES	cs1000-snmp-7.65.16.21-00.i686.000
31	Yes	20/02/14	NO	YES	cs1000-shared-omm-7.65.16.21-2.i386.000
34	Yes	20/02/14	YES	YES	cs1000-ipsec-7.65.16.22-1.i386.000
36	Yes	20/02/14	NO	YES	cs1000-cppmUtil-7.65.16.22-1.i686.000
39	Yes	23/12/14	YES	YES	cs1000-shared-xmsg-7.65.16.22-1.i386.000
40	Yes	23/12/14	NO	YES	cs1000-sps-7.65.16.23-1.i386.000
41	Yes	23/12/14	YES	YES	jdk-1.6.0_81-fcs.i586.000
42	Yes	23/12/14	YES	YES	cs1000-cs-7.65.P.100-03.i386.000
43	Yes	23/12/14	NO	YES	bash-3.2-33.el5_11.4.i386.000
44	Yes	23/12/14	NO	YES	tzdata-2014g-1.el5.i386.000
45	Yes	23/12/14	YES	YES	cs1000-tps-7.65.16.23-7.i386.000
46	Yes	23/12/14	YES	YES	cs1000-vtrk-7.65.16.23-24.i386.000

## 5. Configure Avaya Communication Server 1000

These Application Notes use the Incoming Digit Translation feature to receive calls, the Numbering Plan Area Code (NPA), and the Special Number (SPN) features to route calls from the CS1000 to the PSTN, via SIP trunks to the MTS Allstream SIP Trunk Service network.

These Application Notes assume that the basic CS1000 configuration has already been administered. For further information on CS1000, please consult the references in **Section 11**.

The procedures below describe the configuration details for configuring the CS1000.

### 5.1. Log into Communication Server 1000 System

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

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

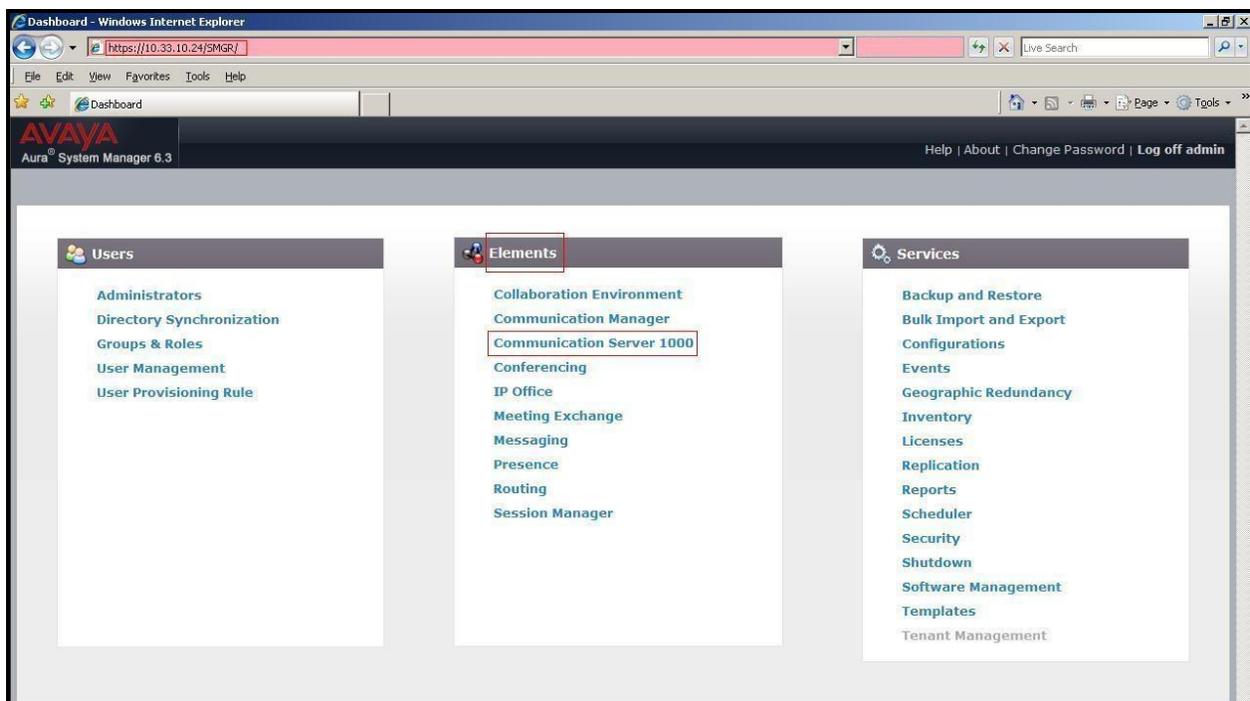


Figure 2 – System Manager Home Screen

The **Avaya Communication Server 1000 Management** screen is displayed. Click on the **Element Name** of the CS1000 Element as highlighted in red box below:

Host Name: 10.33.10.24 User Name: admin

**Elements**

New elements are registered into the security framework, or may be added as simple hyperlinks. Click an element name to launch its management service. You can optionally filter the list by entering a search term.

Search Reset

<input type="checkbox"/>	Element Name	Element Type	Release	Address	Description
<input type="checkbox"/>	smgr.bwdev.com (primary)	Base OS	7.6	10.33.10.24	Base OS element.
<input type="checkbox"/>	EM on car3-sjpl-ucm	CS1000	7.6	10.10.97.96	New element.
<input type="checkbox"/>	car3-cores.bwdev.com (member)	Linux Base	7.6	10.10.97.179	Base OS element.
<input type="checkbox"/>	car3-sjpl-ucm.bwdev.com (member)	Linux Base	7.6	10.10.97.175	Base OS element.
<input type="checkbox"/>	car3-ssn-carrier.bwdev.com (member)	Linux Base	7.6	10.10.97.177	Base OS element.
<input type="checkbox"/>	10.10.97.97	Media Gateway Controller	7.6	10.10.97.97	New element.

**Figure 3 – Communication Server 1000 Management**

Log into the CS1000 using an appropriate **User ID** and **Password**.

AVAYA

Use this page to access the server by IP address. You will need to log in again when switching to another server, even if it is in the same security domain.

Important: Only accounts which have been previously created in the primary security server are allowed. Expired or reset passwords that normally must be changed during login will fail authentication in this mode (use the link to manual password change instead). Local OS-authenticated User IDs cannot be used.

[Go to central login for Single Sign-On](#)

User ID: admin

Password: .....

Log In

[Change Password](#)

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

The CS1000 Element Manager **System Overview** page is displayed as shown in **Figure 5**.

IP Address: 10.10.97.96  
Type: Avaya Communication Server 1000E CPPM Linux  
Version: 4121  
Release: 765 P +



**Figure 5 – Element Manager System Overview**

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

Using Putty, SSH to the IP address of the CS1000 Signaling Server using an account with administrator credentials.

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

login as: ← **Enter an account with administrator credentials**

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

admin@10.10.97.177's password: ← **Enter the password**

Last login: Wed Jan 14 07:20:18 2014 from 10.10.98.78

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

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

USERID? ← **Enter the user account**

PASS? ← **Enter the password**

.

TTY #08 LOGGED IN ADMIN 07:39 01/14/2015

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

>

**Note:** This screen can be used for monitoring of BUG(s), ERROR and AUD messages.

## 5.2. Administer IP Telephony Node

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

### 5.2.1. Obtain Node IP address

These Application Notes assume that the basic CS1000 configuration has already been administered and that a Node has already been created. This section describes the steps for configuring a Node (Node ID 3000) in CS1000 IP network to work with MTS Allstream SIP Trunk Service. For further information on CS1000, please consult the references in **Section 11**.

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

The screenshot displays the AVAYA CS1000 Element Manager interface. The left sidebar shows a navigation tree with 'System' and 'IP Network' expanded, and 'Nodes: Servers, Media Cards' selected. The main content area shows the 'IP Telephony Nodes' configuration page. At the top, it indicates 'Managing: 10.10.97.96 Username: admin' and 'System » IP Network » IP Telephony Nodes'. Below this, there are buttons for 'Add...', 'Import...', 'Export...', and 'Delete', along with 'Print | Refresh'. A table lists the nodes:

<input type="checkbox"/>	Node ID ▲	Components	Enabled Applications	ELAN IP	Node/TLAN IPv4	Node/TLAN IPv6	Status
<input type="checkbox"/>	3000	1	LTPS, Gateway ( SIPGw )	-	10.10.97.178		Synchronized
<input type="checkbox"/>	3002	1	SIP Line, LTPS	-	10.10.97.176		Synchronized

At the bottom, there are checkboxes for 'Show:  Nodes', ' Component servers and cards', and ' IPv6 address'.

**Figure 6 – IP Telephony Node**

The **Node Details** screen is displayed in **Figure 7** with the IP address of the CS1000 node. **Call server IP address: 10.10.97.96**. The **Node IPv4 address 10.10.97.178** is a virtual address which corresponds to the **TLAN IPv4 address 10.10.97.177** of the Signaling Server/SIP Signaling Gateway. The SIP Signaling Gateway uses this Node IP address to communicate with other components to process SIP calls.

**AVAYA CS1000 Element Manager**

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

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

Node ID:  \* (0-9999)

Call server IP address:  \*

TLAN address type:  IPv4 only  
 IPv4 and IPv6

**Embedded LAN (ELAN)**  
Gateway IP address:  \*  
Subnet mask:  \*

**Telephony LAN (TLAN)**  
Node IPv4 address:  \*  
Subnet mask:  \*

Node IPv6 address:

\* Required Value.

**Associated Signaling Servers & Cards**

Select to add    [Print](#) | [Refresh](#)

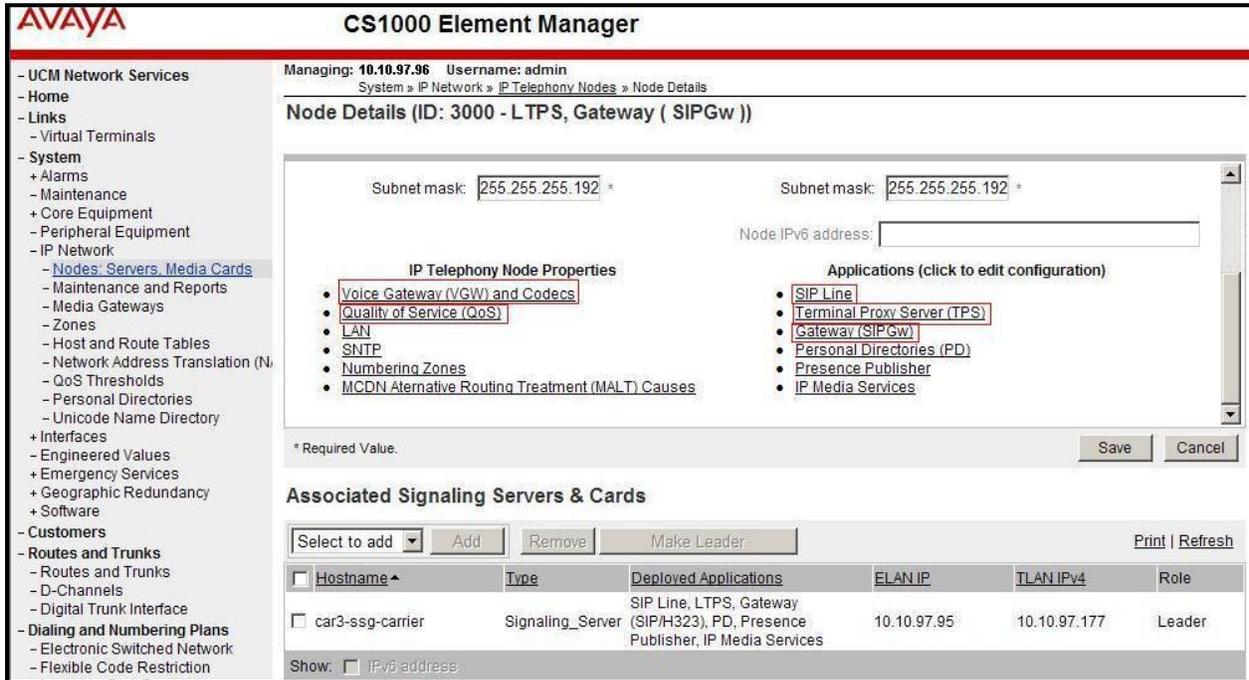
<input type="checkbox"/>	Hostname	Type	Deployed Applications	ELAN IP	TLAN IPv4	Role
<input type="checkbox"/>	car3-ssg-carrier	Signaling_Server	SIP Line, LTPS, Gateway (SIP/H323), PD, Presence Publisher, IP Media Services	10.10.97.95	10.10.97.177	Leader

Show:  IPv6 address

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

**Figure 7 – Node Details 1**

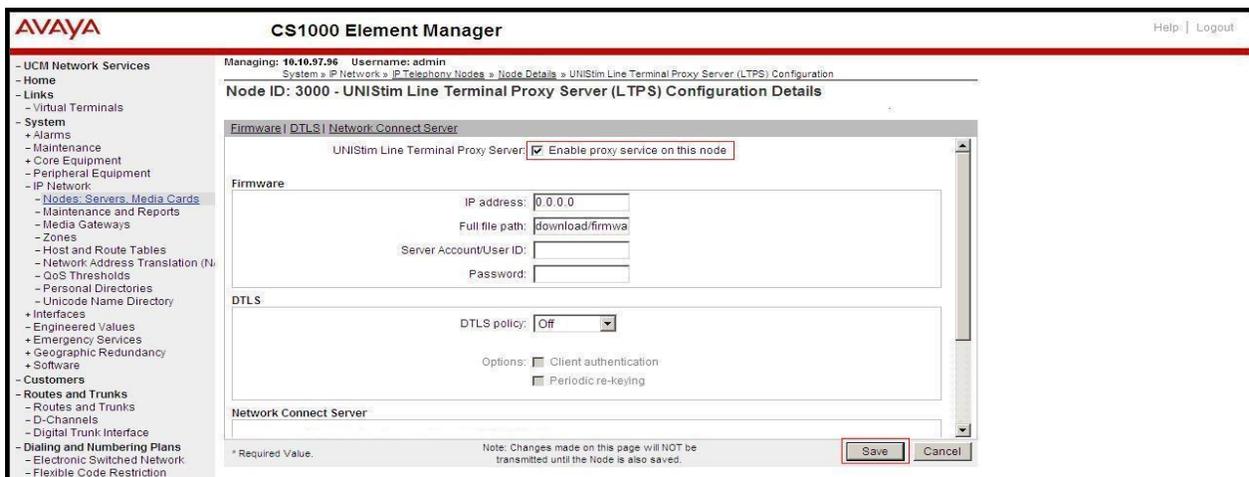
The **Node Details** screen is displayed in **Figure 8** with the IP Telephony Node Properties and Applications.



**Figure 8 – Node Details 2**

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

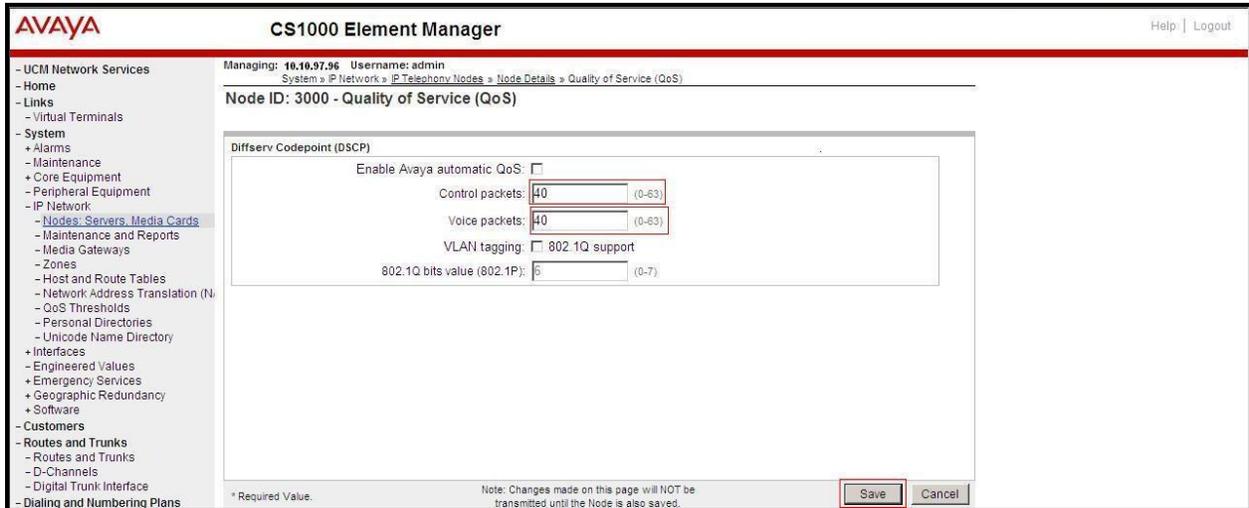
Continuing from **Section 5.2.1**, on the **Node Details** page, select the **Terminal Proxy Server (TPS)** link as shown in **Figure 8**. Check the **UNISim Line Terminal Proxy Server** checkbox to enable proxy service on this node and then click the **Save** button as shown in **Figure 9**.



**Figure 9 – TPS Configuration Details**

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

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



**Figure 10 – QoS Configuration Details**

### 5.2.4. Synchronize New Configuration

Continuing from **Section 5.2.3**, return to the **Node Details** page (**Figure 7**) and click on the **Save** button. The **Node Saved** screen is displayed. Click on **Transfer Now**.



**Figure 11 – Node Saved Screen**

The **Synchronize Configuration Files (Node ID <3000>)** screen is displayed. Check the **car3-ssg-carrier** checkbox and click on **Start Sync**. When the synchronization completes, check the **car3-ssg-carrier** checkbox and click on the **Restart Applications**.

Managing: 10.10.97.96 Username: admin  
System » IP Network » IP Telephony Nodes » Synchronize Configuration Files

**Synchronize Configuration Files (Node ID <3000>)**

Note: Select components to synchronize their configuration files with call server data. This process transfers server INI files to selected components, and requires a restart\* of applications on affected server(s) when complete.

Start Sync Cancel Restart Applications Print Refresh

<input checked="" type="checkbox"/>	Hostname	Type	Applications	Synchronization Status
<input checked="" type="checkbox"/>	car3-ssg-carrier	Signaling_Server	SIP Line, LTPS, Gateway (SIP/H323), PD, Presence Publisher, IP Media Services	Sync required

\* Application restart is only required for initial system configuration or if changes have been made to general LAN configurations, SNTP settings, SIP and H323 Gateway settings, network connectivity related parameters like ports and IP address, enabling or disabling services, or adding or removing application servers.

**Figure 12 – Node Synchronized Screen**

## 5.3. Administer Voice Codec

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

Select **IP Network** → **Nodes: Servers, Media Cards** from the left pane and on the **IP Telephony Nodes** screen displayed (not shown), select the **Node ID** of the CS1000 system. The **Node Details** screen is displayed (see **Section 5.2.1** for more details). On the **Node Details** page shown in **Figure 8**, click on **Voice Gateway (VGW) and Codecs**.

MTS Allstream SIP Trunk Service supports **G.711 a-law**, **G.711 mu-law** and **G.729** with **Voice payload size 20 milliseconds per frame**. Uncheck **Voice Activity Detection (VAD)** checkbox. Click on the **Save** button.

The screenshot displays the AVAYA CS1000 Element Manager interface. The left-hand navigation pane shows a tree structure with 'IP Network' expanded to 'Nodes: Servers, Media Cards'. The main content area is titled 'Node ID: 3000 - Voice Gateway (VGW) and Codecs' and contains three tabs: 'General', 'Voice Codecs', and 'Fax'. The 'Voice Codecs' tab is selected, showing configuration for three codecs: G711, G722, and G729. For each codec, there is a checkbox for 'Enabled', a 'Voice payload size' dropdown menu (all set to 20), and a 'Voice playback (jitter buffer) delay' section with 'Nominal' and 'Maximum' dropdown menus. The 'Voice Activity Detection (VAD)' checkbox is unchecked. At the bottom right, there are 'Save' and 'Cancel' buttons, with the 'Save' button highlighted by a red box. A note at the bottom states: 'Note: Changes made on this page will NOT be transmitted until the Node is also saved.'

**Figure 13 – 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 13**, select **IP Network** → **Media Gateways**. The Media Gateways page will appear (not shown). Click on the **MGC** which is located on the right of the page. In the following screen, scroll down to select the **Codec G711** and **Code G729A** with **Voice payload size 20 ms/frame** and uncheck **VAD** as shown in **Figure 14**. Scroll down to the bottom of the page and click on the **Save** button (not shown).

**AVAYA** CS1000 Element Manager

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

- VGW and IP phone codec profile

Enable echo canceller

Echo canceller tail delay 128 (milliseconds)

Enable dynamic attenuation

Voice activity detection threshold 1 (0 - 4 DBM)

Idle noise level 0 (0 - 1 DBM)

R factor calculation

DTMF tone detection

Enable low latency mode

Remove DTMF delay (squelch DTMF from TDM to IP)

Enable modem/fax pass through mode

Enable V.21 FAX tone detection

Fax TCF method 2

FAX maximum rate 14400 (bps)

FAX playout nominal delay 100 (0 - 300 milliseconds)

FAX no activity timeout 20 (10 - 32000 milliseconds)

FAX packet size 30

- Codec G711 Select

Codec name G711

Voice payload size 20 (ms/frame)

Voice playout (jitter buffer) nominal delay 20

Modifications may cause changes to dependent settings

Voice playout (jitter buffer) maximum delay 40

Modifications may cause changes to dependent settings

VAD

- Codec G729A Select

Codec name G729A

Voice payload size 20 (ms/frame)

Voice playout (jitter buffer) nominal delay 20

**Figure 14 – Media Gateways Configuration Details**

## 5.4. Zones and Bandwidth Management

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

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

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

Select **IP Network** → **Zones** from the left pane (not shown), click on **Bandwidth Zones** as shown in **Figure 15**.

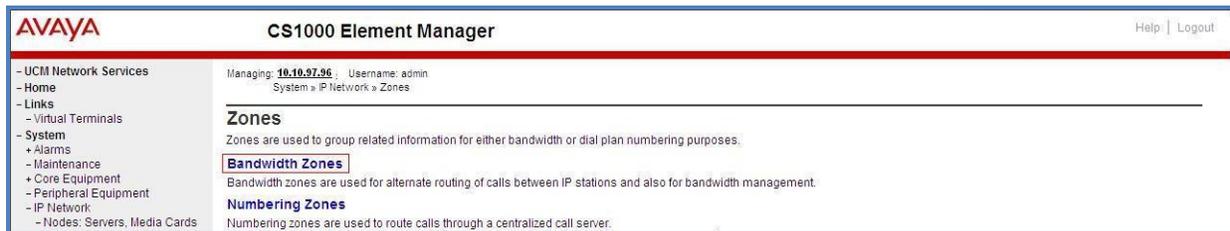


Figure 15 – Zones Page

The **Bandwidth Zones** screen is displayed as shown in **Figure 16**. Click **Add** to create a new zone for IP Phones.

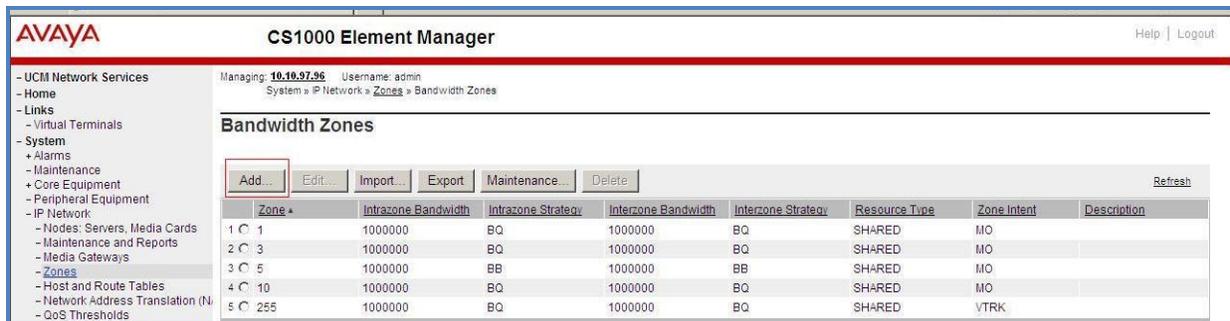


Figure 16 – Bandwidth Zones

Select and input the values as shown below (in the red boxes) in **Figure 17**, and click on the **Submit** button.

- **Intrazone Bandwidth (INTRA\_BW): 1000000.**
- **Intrazone Strategy (INTRA\_STGY):** Set codec for local calls. Select **Best Quality (BQ)** to use G.711 as the first priority codec for negotiation or select **Best Bandwidth (BB)** to use G.729 as the first priority codec for negotiation.
- **Interzone Bandwidth (INTER\_BW): 1000000.**
- **Interzone Strategy (INTER\_STGY):** Set codec for the calls over trunk. Select **Best Quality (BQ)** to use G.711 as the first priority codec for negotiation or select **Best Bandwidth (BB)** to use G.729 as the first priority codec for negotiation.
- **Zone Intent (ZBRN):** Select **MO (MO)** for IP phones, and VGW.

The screenshot shows the 'Zone Basic Property and Bandwidth Management' configuration page for Zone 10. The form contains the following fields and values:

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

**Figure 17 – Bandwidth Management Configuration Details – IP phone**

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

Follow the steps described in **Section 5.4.1** to create a zone for the virtual SIP trunk. The difference is in the **Zone Intent (ZBRN)** field. Select **VTRK** for virtual trunk as shown in **Figure 18** and then click on the **Submit** button.

The screenshot shows the 'Zone Basic Property and Bandwidth Management' configuration page for Zone 255. The form contains the following fields and values:

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

**Figure 18 – Bandwidth Management Configuration Details – Virtual SIP trunk**

## 5.5. Administer SIP Trunk Gateway

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

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

Select **Customers** in the left pane. The **Customers** screen is displayed. Click on the link associated with the appropriate customer, in this case **00**.

Customer Number	Total Routes	Total Trunks
1 00	7	125
2 01	0	0

Figure 19 – Customer – ISDN Configuration 1

The system can support more than one customer with different network settings and options. The **Customer Details** page will appear. Select the **Feature Packages** option from **Customer Details** page.

Customer Details
Basic Configuration
Application Module Link
Attendant
Call Detail Recording
Call Party Name Display
Call Redirection
Centralized Attendant Service
Controlled Class of Service
Features
<b>Feature Packages</b>
Flexible Feature Codes
Intercept Treatments
ISDN and ESN Networking
Listed Directory Numbers
Media Services Properties
Mobile Service Directory Numbers
Multi-Party Operations
Night Service
Recorded Overflow Announcement
SIP Line Service
Timers

Figure 20 – Customer – ISDN Configuration 2

The screen is updated with a listing of available **Feature Packages** (not all features are shown in **Figure 21** below). Select **Integrated Services Digital Network** to edit the parameters shown below. Check the **Integrated Services Digital Network** option, and retain the default values for all remaining fields. Scroll down to the bottom of the screen, and click on the **Save** button (not shown).

**AVAYA CS1000 Element Manager**

Managing: 10.10.97.96 Username: admin  
 Customers » Customer 00 » Customer Details » Feature Packages

**Feature Packages**

+ Do Not Disturb Individual	Package: 9
+ End-to-End Signaling	Package: 10
+ Message Waiting Center	Package: 46
+ New Flexible Code Restriction	Package: 49
+ Set Relocation	Package: 53
+ Network Alternate Route Selection	Package: 58
+ Distinctive Ringing	Package: 74
+ Departmental Listed Directory Number	Package: 76
+ Command Status Link	Package: 77
+ Pretranslation	Package: 92
+ Dialed Number Identification System	Package: 98
+ Malicious Call Trace	Package: 107
+ Incoming Digit Conversion	Package: 113
+ Directed Call Pickup	Package: 115
+ Enhanced Music	Package: 119
+ Station Camp-On	Package: 121
+ Integrated Digital Access	Package: 122
+ Digital Private Network Signaling System 1	Package: 123
+ Flexible Tones and Cadences	Package: 125
+ Multifrequency Compelled Signaling	Package: 128
+ International Supplementary Features	Package: 131
+ Enhanced Night Service	Package: 133
<b>- Integrated Services Digital Network</b>	<b>Package: 145</b>

+ Dial Access Prefix on CLID table entry option

Integrated Services Digital Network:

- Virtual private network identifier:  (1 - 16383)

- Private network identifier:  (1 - 16383)

- Node DN:

Multi-location business group:  (0 - 65535)

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

**Figure 21 – Customer – ISDN Configuration 3**

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

Select **IP Network** → **Nodes: Servers, Media Cards** from the left pane. In the **IP Telephony Nodes** screen displayed (not shown), select the **Node ID** of the CS1000 system. The **Node Details** screen is displayed as shown in **Figure 8, Section 5.2.1**.

On the **Node Details** screen, select **Gateway (SIPGw)**. Under the **General** tab of the **Virtual Trunk Gateway Configuration Details** screen, enter the following values (highlighted in red boxes) for the specified fields, and retain the default values for the remaining fields as shown in **Figure 22**. The **SIP domain name** and **Local SIP port** should be matched in the configuration of Avaya SBCE (in **Section 6.2**, and **6.6**).

The screenshot displays the 'AVAYA CS1000 Element Manager' interface. The main content area is titled 'Node ID: 3000 - Virtual Trunk Gateway Configuration Details'. The 'General' tab is selected, showing the following configuration fields:

- Vtrk gateway application:  Enable gateway service on this node
- Vtrk gateway application: SIP Gateway (SIPGw) (highlighted in red)
- SIP domain name: bwdev7.com (highlighted in red)
- Local SIP port: 5060 (highlighted in red) \*(1 - 65535)
- Gateway endpoint name: car3-ssg-carrier (highlighted in red)
- Gateway password: (empty)
- Application node ID: 3000 (highlighted in red) \*(0-9999)
- Enable failsafe NRS:

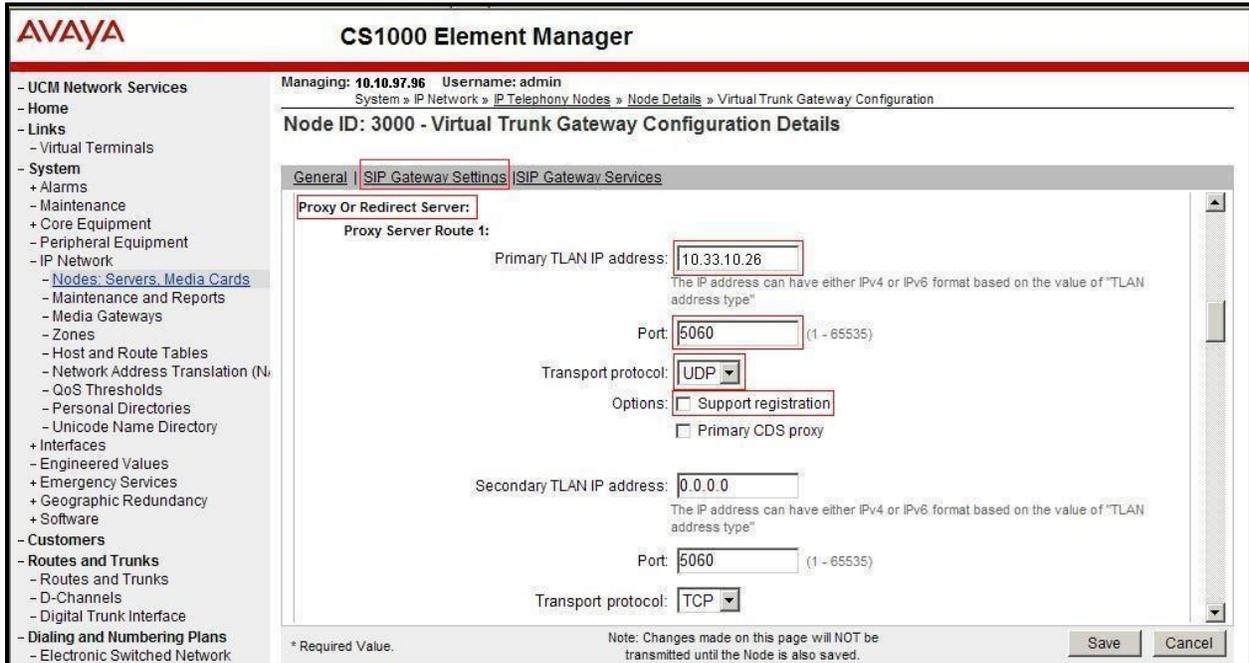
Additional fields and sections include:

- Virtual Trunk Network Health Monitor:  Monitor IP addresses (listed below)
- Monitor IP: (empty) Add
- Monitor addresses: (empty) Remove

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

**Figure 22 – 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 23**. Enter the IP address of Session Manager in the **Primary TLAN IP address** field. Enter **5060** for **Port** and select **UDP** for **Transport protocol**. This should be matched in the configuration of Session Manager (see to **Section 6.5.1**). Uncheck the **Support registration** checkbox.



**Figure 23 – Virtual Trunk Gateway Configuration Details**

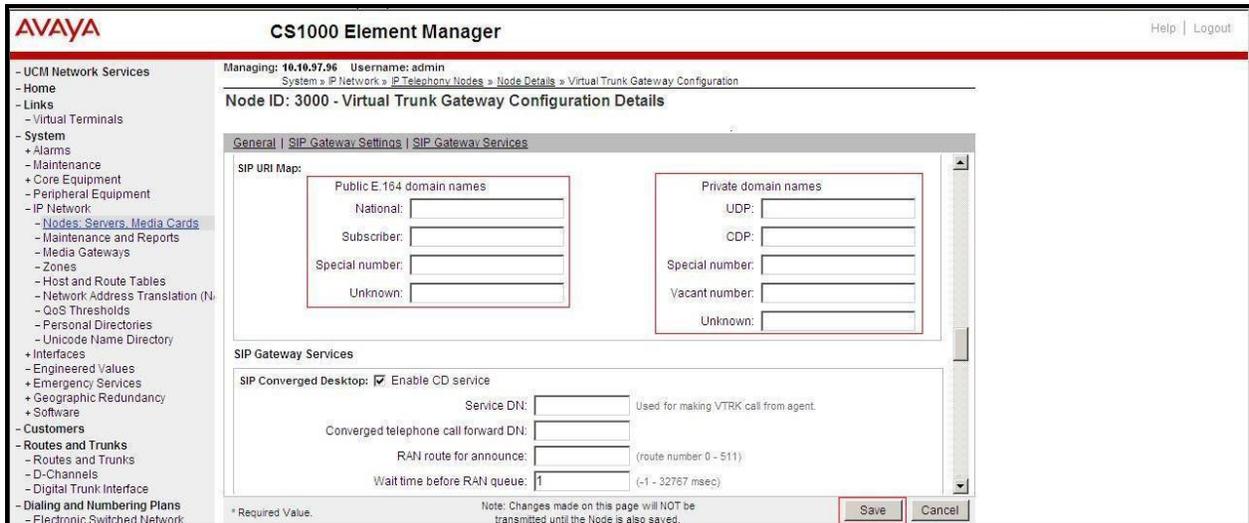
On the same page as shown in **Figure 23**, scroll down to the **SIP URI Map** section. Under **Public E.164 domain names**, enter the following:

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

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

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

The remaining fields can be left at their default values as shown in **Figure 24**. Click on the **Save** button.

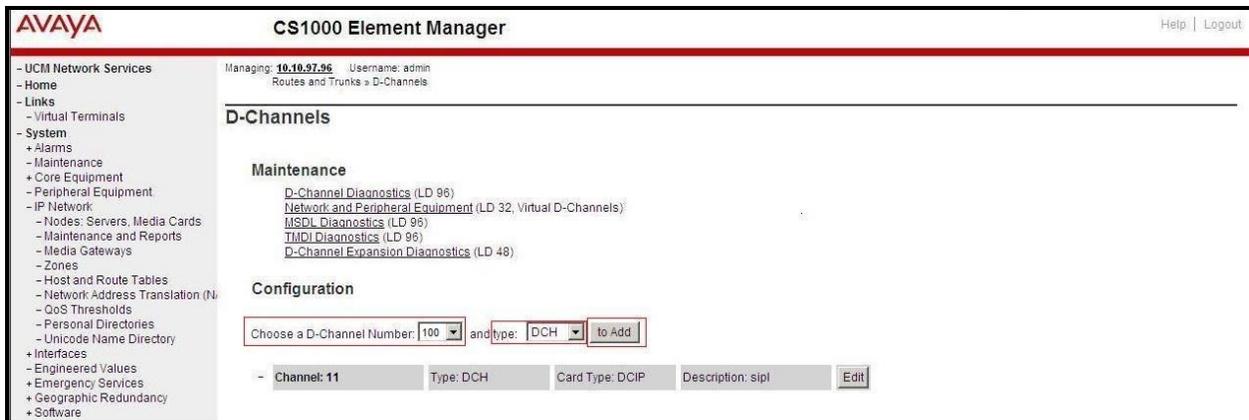


**Figure 24 – Virtual Trunk Gateway Configuration Details**

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 25**. Click on the **to Add** button.



**Figure 25 – D-Channels**

The **D-Channels 100 Property Configuration** screen is displayed next, as shown in **Figure 26**. 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 26**. Other fields are left as default.

The screenshot displays the 'D-Channels 100 Property Configuration' screen in the AVAYA CS1000 Element Manager. The interface is divided into a left-hand navigation menu and a main configuration area. The main area is titled '- Basic Configuration' and contains a table with 'Input Description' and 'Input Value' columns. The configuration fields are as follows:

Input Description	Input Value
Action Device And Number (ADAN):	DCH
D channel Card Type:	DCIP
Designator:	VoIP
Recovery to Primary:	<input type="checkbox"/>
PRI loop number for Backup D-channel:	
User:	Integrated Services Signaling Link Dedicated (ISLD)
Interface type for D-channel:	Meridian Meridian1 (SL1)
Country:	ETS 300 =102 basic protocol (ETSI)
D-Channel PRI loop number:	
Primary Rate Interface:	<input type="text"/> more PRI
Secondary PRI2 loops:	
Meridian 1 node type:	Slave to the controller (USR)
Release ID of the switch at the far end:	25
Central Office switch type:	100% compatible with Bellcore standard (STD)
Integrated Services Signaling Link Maximum:	4000 (Range: 1 - 4000)
Signalling server resource capacity:	1800 (Range: 0 - 3700)
<b>Basic options (BSCOPT)</b>	
<b>Advanced options (ADVOPT)</b>	
- Layer 3 call control message count per 5 second time interval:	300 (Range: 60 - 350)
- Number of Status Enquiry Messages sent within 128 ms:	1
- Map channel number to timeslots on a PRI2 loop:	<input checked="" type="checkbox"/>
<b>H323 Overlap Signaling Settings (H323)</b>	
- Overlap Receiving:	<input type="checkbox"/>
- Overlap Sending:	<input type="checkbox"/>
--Overlap Timer:	<input type="text"/>
- Multilocation Business Group Allowed:	<input type="checkbox"/>
- Network Attendant Service Allowed:	<input checked="" type="checkbox"/>
<b>Link Access Protocol for D-channel (LAPD)</b>	
<b>Feature Packages</b>	

**Figure 26 – D-Channel Configuration**

Click on **Basic Options (BSCOPT)** and click on the **Edit** button on the **Remote Capabilities** field as shown in **Figures 27**.

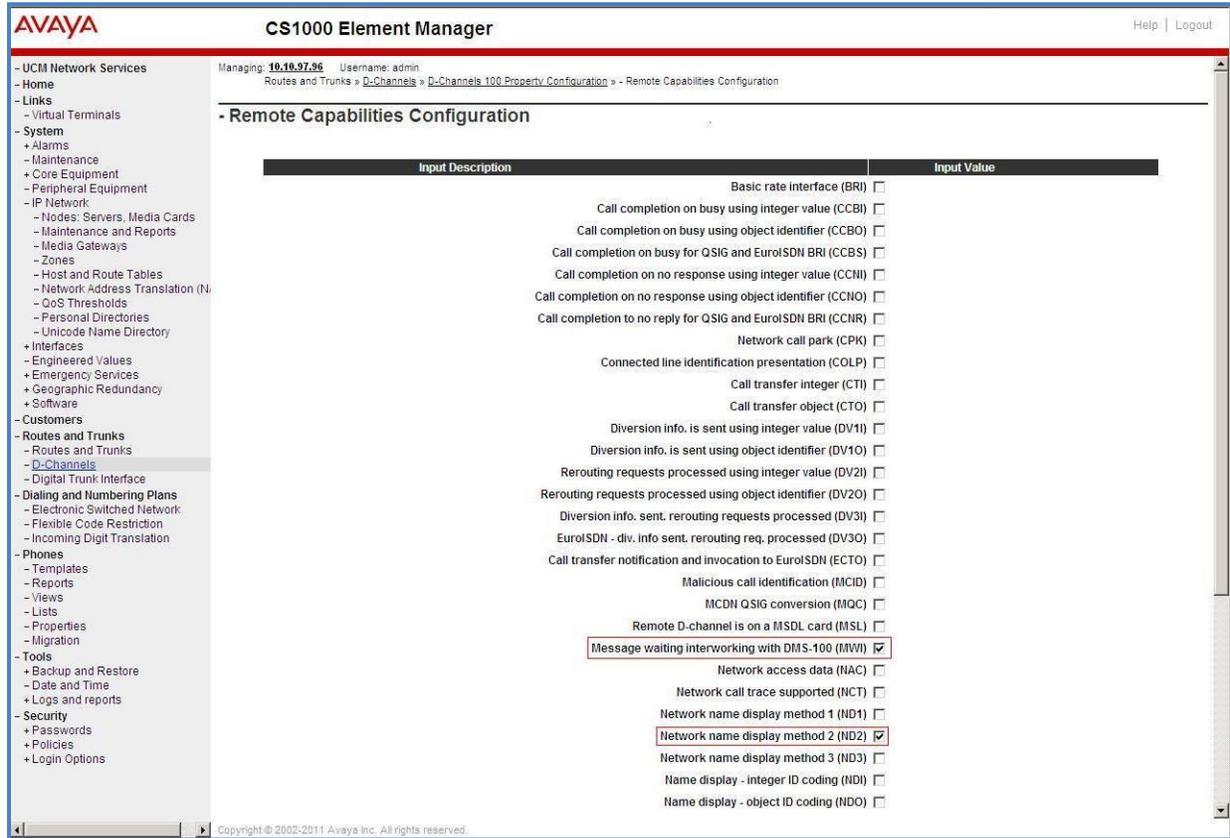
The screenshot displays the AVAYA CS1000 Element Manager interface. The left-hand navigation menu is expanded to show the 'D-Channels' section, with 'Basic options (BSCOPT)' highlighted. The main configuration area contains the following fields and options:

- Action Device And Number (ADAN): DCH
- D channel Card Type: DCIP
- Designator: VoIP
- Recovery to Primary:
- PRI loop number for Backup D-channel:
- User: Integrated Services Signaling Link Dedicated (ISLD)
- Interface type for D-channel: Meridian Meridian1 (SL1)
- Country: ETS 300 =102 basic protocol (ETSI)
- D-Channel PRI loop number:
- Primary Rate Interface:  more PRI
- Secondary PRI2 loops:
- Meridian 1 node type: Slave to the controller (USR)
- Release ID of the switch at the far end: 25
- Central Office switch type: 100% compatible with Bellcore standard (STD)
- Integrated Services Signaling Link Maximum: 4000 (Range: 1 - 4000)
- Signalling server resource capacity: 1800 (Range: 0 - 3700)
- Primary D-channel for a backup DCH:  (Range: 0 - 254)
- PINX customer number:
- Progress signal:
- Calling Line Identification:
- Output request Buffers: 32
- D-channel transmission Rate: 56 kb/s when LCMT is AMI (56K)
- Channel Negotiation option: No alternative acceptable, exclusive. (1)
- Remote Capabilities: **Edit**
- B channel Service messaging:

At the bottom of the configuration area, there are buttons for 'Submit', 'Refresh', 'Delete', and 'Cancel'. The footer of the page reads 'Copyright © 2002-2011 Avaya Inc. All rights reserved.'

**Figure 27 – D-Channel Configuration**

The **Remote Capabilities Configuration** page appears as shown in **Figures 28**. Check the **ND2** and the **MWI** checkboxes.



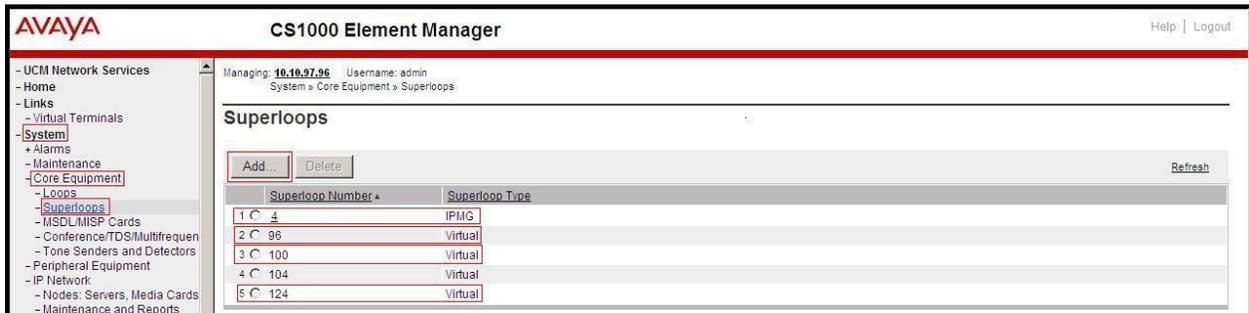
**Figure 28 – Remote Capabilities Configuration**

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

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

#### 5.5.4. Administer Virtual Super-Loop

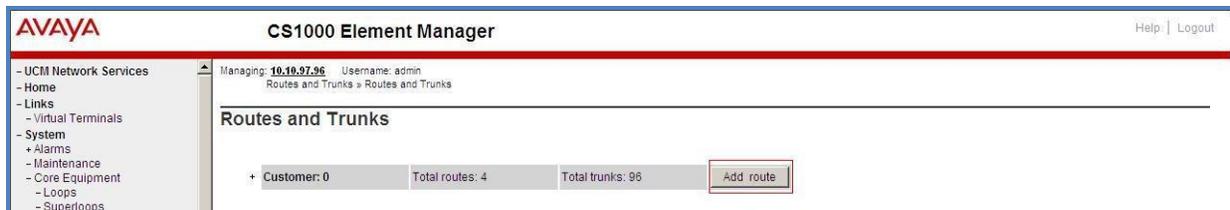
Select **System** → **Core Equipment** → **Superloops** from the left pane to display the **Superloops** screen. If the Superloop does not exist, please click the **Add** button to create a new one as shown in **Figure 29**. In this example, Superloop 4, 96, 100, and 124 have been added and are being used.



**Figure 29 – 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 30**.

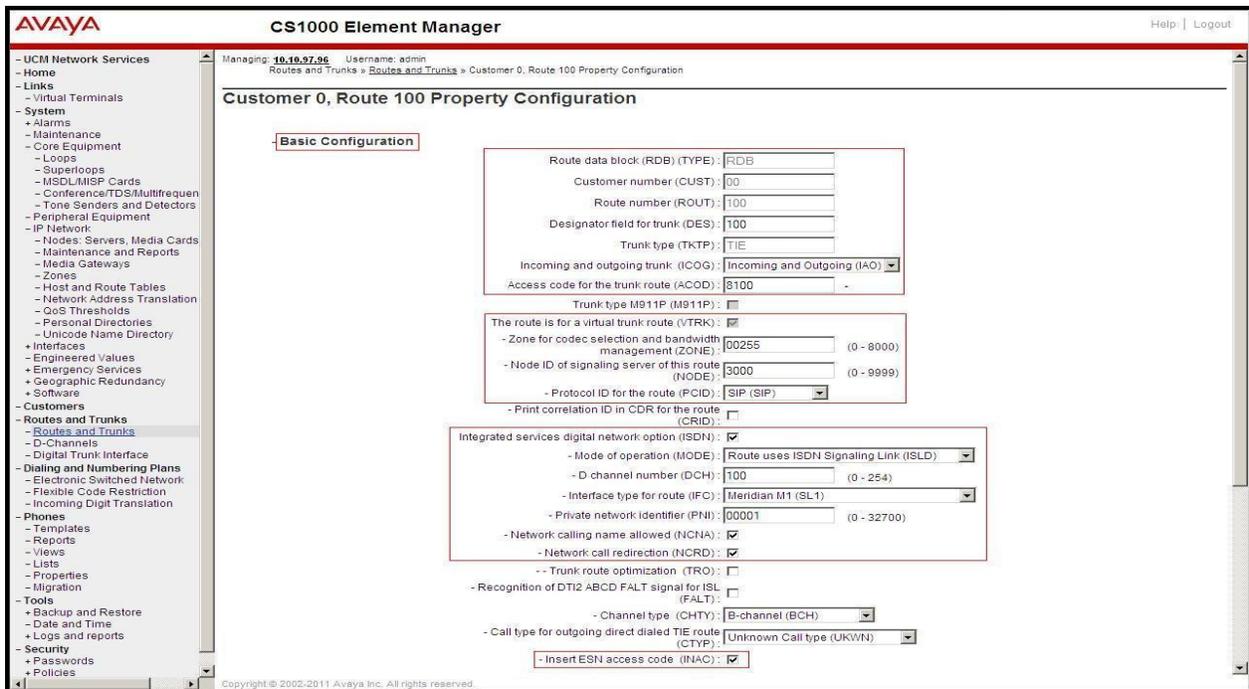


**Figure 30 – Add route**

The **Customer 0, New Route Configuration** screen is displayed next (not shown). The **Basic Configuration** section is displayed. Enter the following values for the specific fields, and retain the default values for the remaining fields. The screenshot of Basic Configuration section of existing route 100 is displayed to edit as shown in **Figures 31**.

- **Route data block (RDB) (TYPE):** **RDB** as default.
- **Customer number (CUST):** **0** as customer 0 is in used.
- **Route number (ROUT):** Enter an available route number (example: route **100**).
- **Designator field for trunk (DES):** A descriptive text (**100**).
- **Trunk type (TKTP):** TIE trunk data block (**TIE**).
- **Incoming and outgoing trunk (ICOG):** **Incoming and Outgoing (IAO)**.
- **Access code for the trunk route (ACOD):** An available access code (example: **8100**).

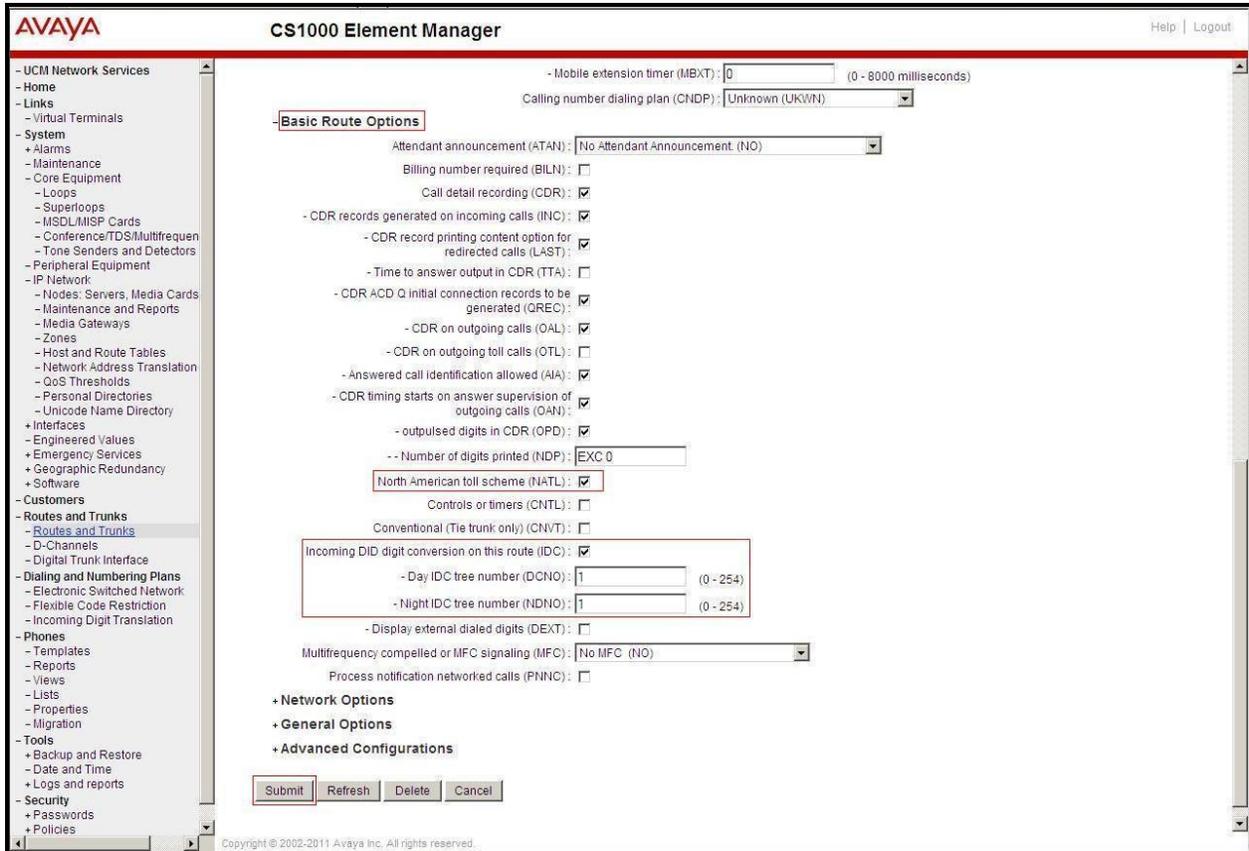
- Check the **The route is for a virtual trunk route (VTRK)** field, to enable four additional fields to appear.
- For the **Zone for codec selection and bandwidth management (ZONE)** field, enter **255** (created in Section 5.4.2). **Note:** The Zone value is filled out as 255, but after it is added, the screen is displayed with prefix 00.
- For the **Node ID of signaling server of this route (NODE)** field, enter the node number **3000** (created in Section 5.2.1).
- Select **SIP (SIP)** from the drop-down list for the **Protocol ID for the route (PCID)** field.
- Check the **Integrated Services Digital Network option (ISDN)** checkbox to enable additional fields to appear. Scrolling down to the bottom of the screen, enter the following values for the specified fields, and retain the default values for the remaining fields.
  - **Mode of operation (MODE):** Select **Route uses ISDN Signaling Link (ISLD)**.
  - **D channel number (DCH):** Enter **100** (created in Section 5.5.3).
  - **Interface type for route (IFC):** Select **Meridian M1 (SL1)**.
  - **Private network identifier (PNI):** Enter **1**. **Note:** The value is filled out as 1, but after it is added, the screen is displayed with prefix 0000.
  - **Network calling name allowed (NCNA):** Check this option to allow calling name display.
  - **Network call redirection (NCRD):** Check this option to allow call redirection.
  - **Insert ESN access code (INAC):** Check this option to insert ESN access code (Refer to Section 5.6.1).



**Figure 31 – Route Configuration 1**

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 32**.

Click on the **Submit** button.



**Figure 32 – Route Configuration 2**

### 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, Route 100 was being added. Click on the **Add trunk** button as shown in **Figure 33**.



**Figure 33 – Routes and Trunks**

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. 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 34**.

**Note:** The Multiple trunk input number (MTINPUT) field may be used to add multiple trunks in a single operation, or repeat the operation for each trunk. In the sample configuration, 32 trunks were created.

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

The screenshot shows the AVAYA CS1000 Element Manager interface. The main title is "Customer 0, Route 100, Trunk 1 Property Configuration". The left sidebar contains a navigation menu with categories like "UCM Network Services", "System", "IP Network", "Customers", and "Routes and Trunks". The "Routes and Trunks" category is expanded, showing "Routes and Trunks" as the selected item. The main content area is titled "Basic Configuration" and contains the following fields:

- Auto increment member number:
- Trunk data block:
- Terminal number:
- Designator field for trunk:
- Extended trunk:
- Member number:  \*
- Level 3 Signaling:
- Card density:
- Start arrangement Incoming:
- Start arrangement Outgoing:
- Trunk group access restriction:
- Channel ID for this trunk:

At the bottom of the "Basic Configuration" section, there is a "Class of Service:" label followed by an "Edit" button. Below this, there is an "Advanced Trunk Configurations" section which is currently empty. At the bottom right of the form, there are three buttons: "Save", "Delete", and "Cancel".

**Figure 34 – New Trunk Configuration**

For **Media Security**, select **Media Security Never (MSNV)**. Enter the values for the specified fields as shown in **Figure 35**. Scroll down to the bottom of the screen and click **Return Class of Service** and click on the **Save** button (shown in **Figure 34**).

The screenshot displays the 'Class of Service' configuration page in the AVAYA CS1000 Element Manager. The interface includes a left-hand navigation menu with categories such as 'UCM Network Services', 'System', 'Interfaces', 'Customers', 'Routes and Trunks', 'Dialing and Numbering Plans', 'Phones', and 'Tools'. The main content area is titled '- Class of Service' and contains a table with two columns: 'Input Description' and 'Input Value'. The table lists various configuration parameters, each with a corresponding dropdown menu. The 'Media Security' parameter is highlighted with a red box, and its value is 'Media Security Never (MSNV)'. Other parameters include 'ACD Priority' (ACD Priority not required (APN)), 'Analog Semi-Permanent Connections' (Analog Semi-Permanent Connections Denied (SPCD)), 'Centrex Switchhook Flash' (Centrex Switchhook Flash Denied (THFD)), and 'Transmission Class of Service' (Non-Transmission Compensated (NTC)). At the bottom of the page, there are two buttons: 'Return Class of Service' and 'Cancel'. The footer of the page contains the text 'Copyright © 2002-2011 Avaya Inc. All rights reserved.'

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

**Figure 35 – Class of Service Configuration**

## 5.5.7. Administer Calling Line Identification Entries

Select **Customers** on the left pane, then select **00 → ISDN and ESN Networking** (Not shown). Click on **Calling Line Identification Entries** as shown in **Figure 36**.

AVAYA CS1000 Element Manager

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

### ISDN and ESN Networking

**General Properties**

Flexible trunk to trunk connection option:

Flexible orbiting prevention timer:

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

National access code:

International access code:

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

Network option:  Coordinated dialing plan routing

Integrated services digital network:

Microsoft converged office dialing plan:

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

**Calling Line Identification**

Information for incoming/outgoing calls:

Size:  (0 - 4000)

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

Figure 36 – ISDN and ESN Networking

Click on **Add** as shown in **Figure 37**.

AVAYA CS1000 Element Manager

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

### Calling Line Identification Entries

Search for CLID

Start range:

End range:

'End range' should not exceed the CLID size specified

Calling Line Identification Entries

Figure 37 – Calling Line Identification Entries

The add entry **0** screen is displayed. Enter or select the following values for the specified fields and retain the default values for the remaining fields. The **Edit Calling Line Identification** of the existing entry **0** is displayed as shown in **Figure 38**.

- **National Code:** Leave it blank.
- **Local Code:** Input prefix digits assigned by MTS Allstream SIP Trunk Service, in this case 6 digits – **647XXX**. This **Local Code** will be used for call display purpose for Call Type = Unknown.
- **Home Location Code:** Input the prefix digits assigned by MTS Allstream SIP Trunk Service, in this case 6 digits – **647XXX**. This **Home Location Code** will be used for call display purpose for Call Type = National (NPA).
- **Local Steering Code:** Input prefix digits assigned by MTS Allstream SIP Trunk Service, in this case 6 digits – **647XXX**. This **Local Steering Code** will be used for call display purpose for Call Type = Local Subscriber (NXX).
- **Use DN as DID:** **YES**.
- **Calling Party Name Display:** Uncheck **Roman characters**.

Click on the **Save** button as shown in **Figure 38**.

The screenshot shows the 'Edit Calling Line Identification 0' configuration page in the AVAYA CS1000 Element Manager. The page is divided into several sections:

- General Properties:**
  - National Code: (0 - 999999)
  - Local Code: 647XXX (1-12 digits)
  - Home Location Code: 647XXX (1-7 digits)
  - Local Steering Code: 647XXX (1-7 digits)
  - Use DN as DID: YES
- Emergency Services Access:**
  - Emergency Local Code: (1-12 digits)
  - Emergency Options:
    - Home national number for emergency services access calls
    - Append the originating directory number for emergency services access calls
- Calling Party Name Display:**
  - Roman characters:
  - CPND Name: (first name, last name)
  - Expected Length: 12
  - Display Format: First name, Last name

At the bottom right, there are 'Save' and 'Cancel' buttons.

**Figure 38 – Edit Calling Line Identification 0**

### 5.5.8. Enable External Trunk to Trunk Transfer

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

Log in to Call Server Overlay CLI (please refer to **Section 5.1.2** for more details).  
Allow External Trunk to Trunk Transfer for Customer Data Block by using **ld 15**.

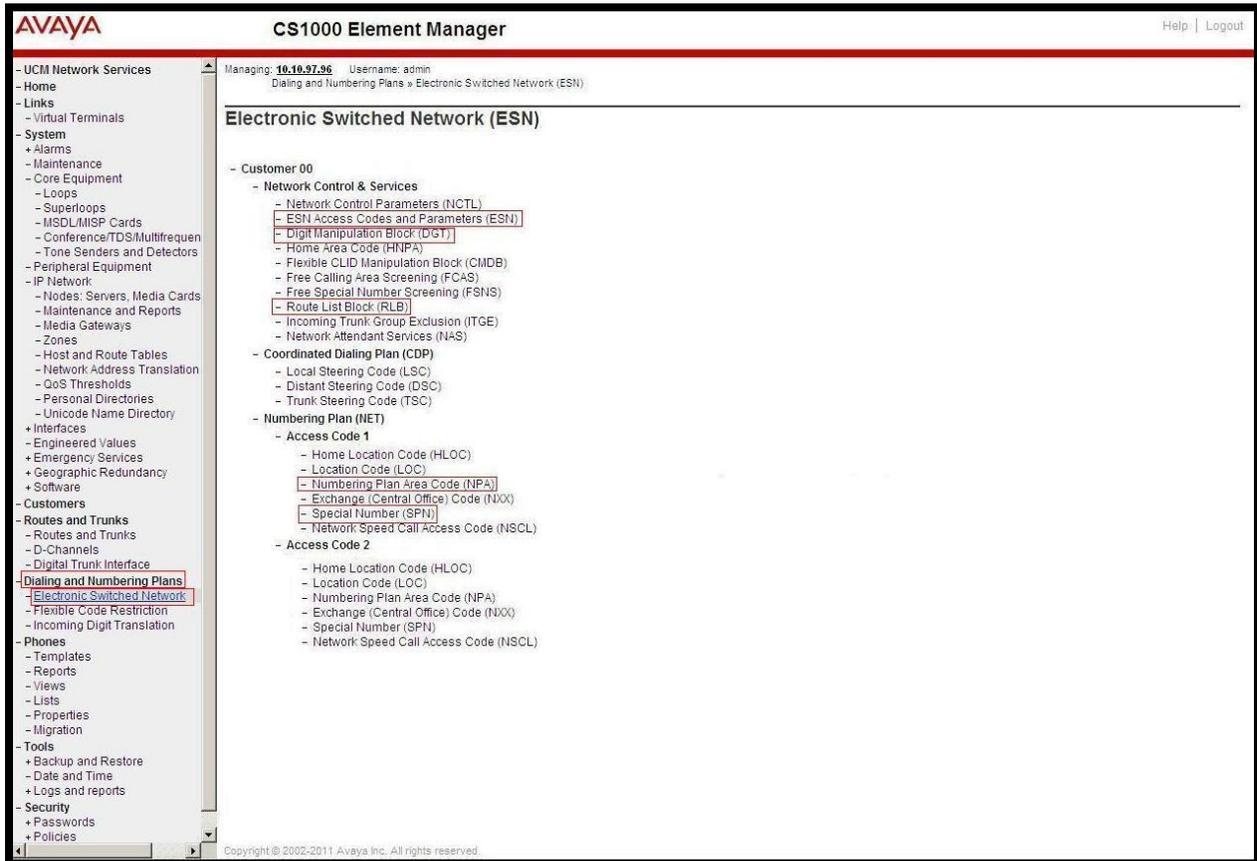
```
>ld 15
CDB000
MEM AVAIL: (U/P): 33600126  USED U P: 8345621 954062  TOT: 45579868
DISK SPACE NEEDED: 1722 KBYTES
REQ: chg
TYPE: net

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

## 5.6. Administer Dialing Plans

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

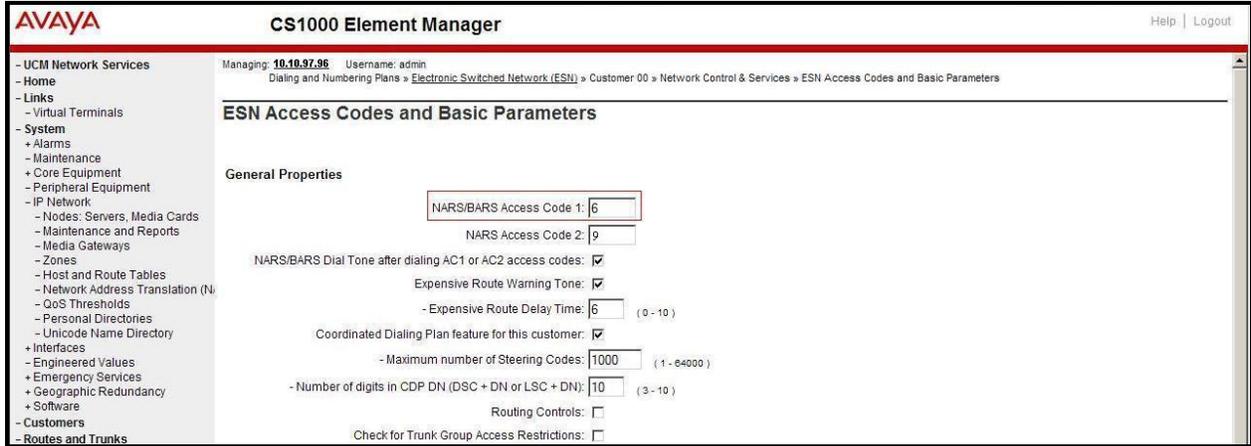
Select **Dialing and Numbering Plans** → **Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen as shown in **Figure 39**.



**Figure 39 – ESN Configuration**

On **Electronic Switched Network (ESN)** screen, select **ESN Access Codes and Parameters** to define **NARS/BARS Access Code 1** as shown in **Figure 40**.

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



**Figure 40 – ESN Access Codes and Parameters**

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

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

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

TYPE NET_DATA
CUST 0
OPT
AC2 xNPA xSPN ← Set NPA, SPN not to associate to ESN Access Code 2
FNP
CLID
...
```

Verify Customer Net Data block by using **ld 21**.

```
>ld 21
PT1000

REQ: prt
TYPE: net
TYPE NET_DATA
CUST 0

TYPE NET_DATA
CUST 00
OPT RTA
AC1 INTL NPA SPN NXX LOC ← NPA, SPN are associated to ESN Access Code 1
AC2
FNP YES
...
```

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

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

Select **Dialing and Numbering Plans** → **Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen as shown in **Figure 39**. Select **Digit Manipulation Block (DGT)**. The **Digit Manipulation Block List** is displayed as shown in **Figure 41**. In the **Please choose the** field, select an available **Digit Manipulation Block Index** from the drop-down list, and click on the **to Add** button.



**Figure 41 – Add a DMI**

The DMI\_14 screen will open. In this testing, no leading digits are to be deleted, therefore, enter **0** for **Number of leading digits to be deleted** and select **NPA (NPA)** for **Call Type to be used by the manipulated digits** and then click on the **Submit** button as shown in **Figure 42**.

AVAYA CS1000 Element Manager

Managing: 10.10.97.96 Username: admin  
Dialing and Numbering Plans > Electronic Switched Network (ESN) > Customer 00 > Network Control & Services > Digit Manipulation Block List > Digit Manipulation Block

Digit Manipulation Block

Digit Manipulation Index numbers: 14

Number of leading digits to be deleted: 0 (0 - 19)

Insert: [ ]

IP Special Number: [ ]

Call Type to be used by the manipulated digits: NPA (NPA)

Submit Refresh Delete Cancel

**Figure 42 – DMI\_14 Configuration**

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

This section shows how to add a RLB associated with the DMI created in **Section 5.6.3**. Select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen as shown in **Figure 39**. Select **Route List Block (RLB)**.

Enter an available value in the textbox for the **Please enter a route list index** (in this case **14**) and click on the **to Add** button as shown in **Figure 43**. The screen shown in **Figure 44** will open.

AVAYA CS1000 Element Manager

Managing: 10.10.97.96 Username: admin  
Dialing and Numbering Plans > Electronic Switched Network (ESN) > Customer 00 > Network Control & Services > Route List Blocks

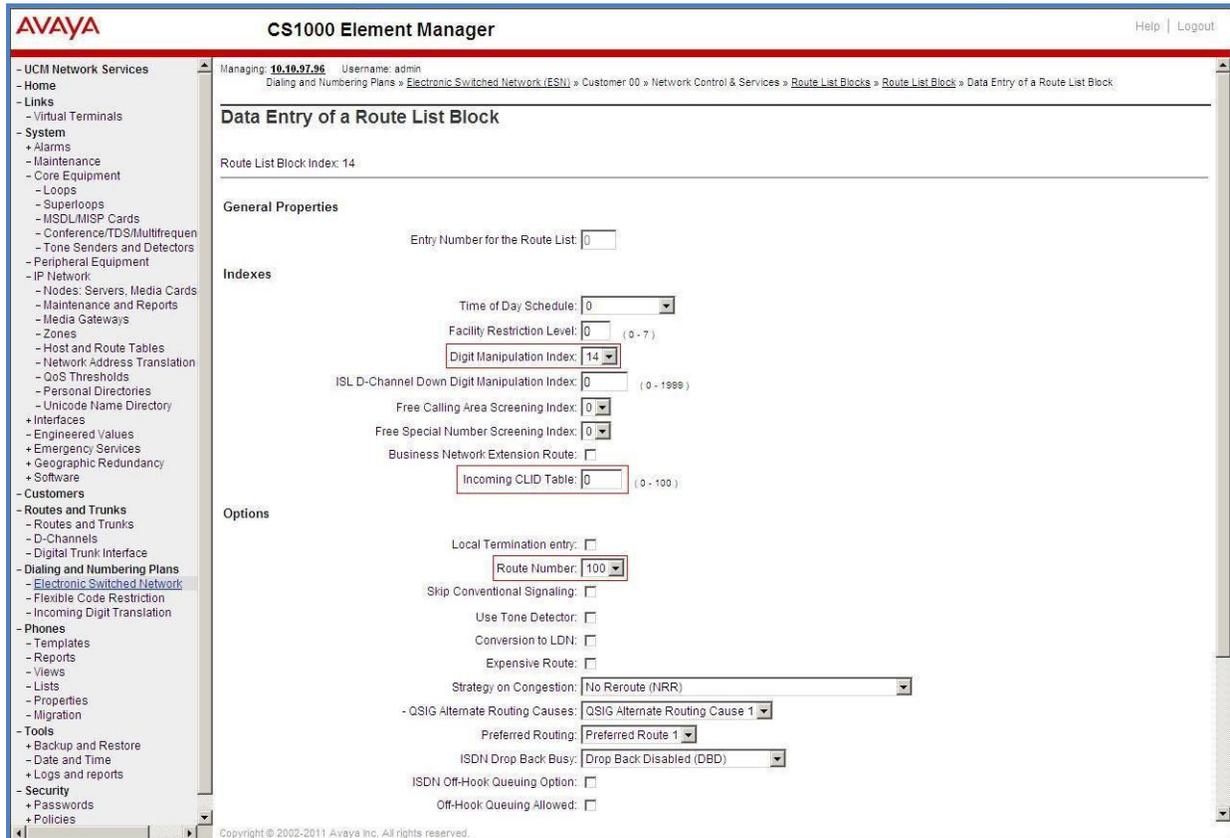
Route List Blocks

Please enter a route list index: 14 (0 - 1999) to Add

**Figure 43 – Add a Route List Block**

Enter the following values for the specified fields, and retain the default values for the remaining fields as shown in **Figure 44**. Scroll down to the bottom of the screen, and click on the **Submit** button (not shown).

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

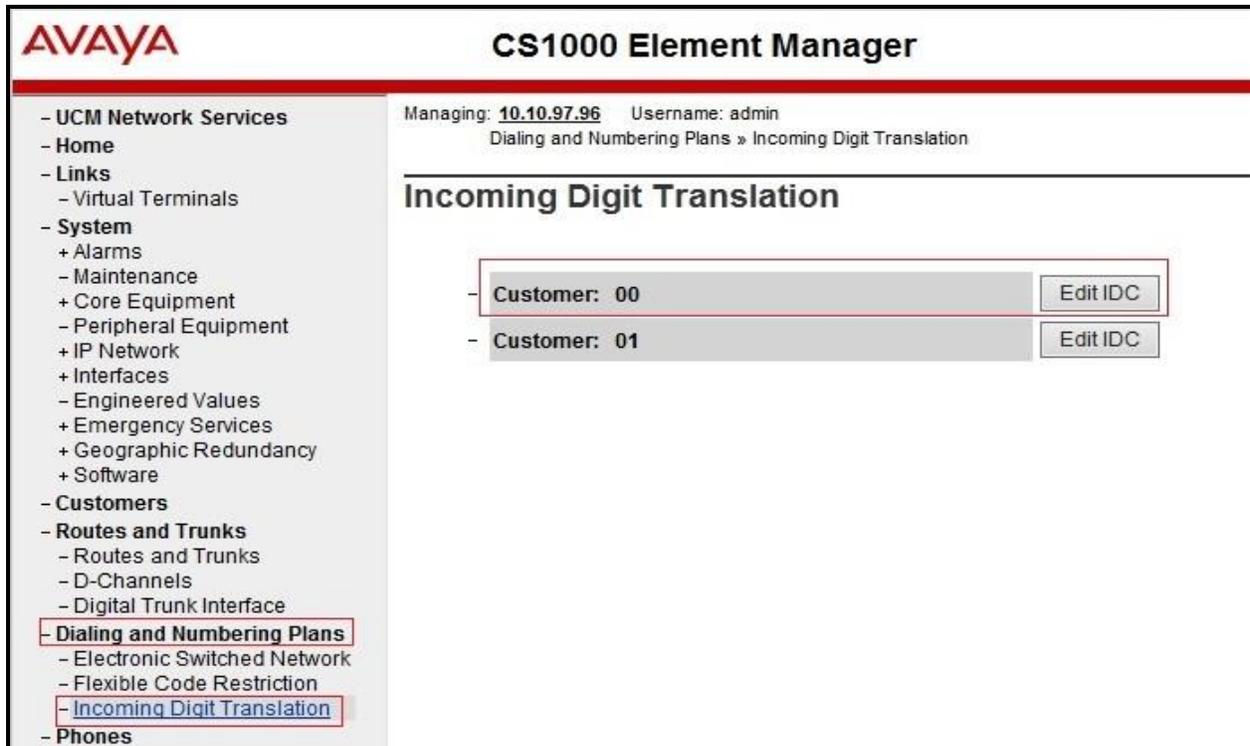


**Figure 44 – RLB\_14 Route List Block Configuration**

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

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

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 45**.



**Figure 45 – Incoming Digit Translation**

Click on the **New DCNO** to create the digit translation mapping. In this example, **Digit Conversion Tree Number 1** has been previously created as shown in **Figure 46**.



**Figure 46 – Incoming Digit Conversion Property**

Detailed configuration of the Digit Conversion Tree Configuration is shown in **Figure 47**. The **Incoming Digits** can be added to map to the Converted Digits which would be the associated CS1000 system phone DN. This **DCNO** has been assigned to route 100 as shown in **Figure 32**.

In the following configuration, the incoming call from the PSTN to DID with prefix **647XXX** will be translated to the associated DN with 4 digits. For testing purposes, DID number **647XXX1264** is translated to **1700** for voicemail testing or translated to 1264 for Mobile Service Access DN number.

Managing: 10.10.97.96 Username: admin  
Dialing and Numbering Plans » Incoming Digit Translation » Customer 00 » Digit Conversion Tree 1 Configuration

### Digit Conversion Tree 1 Configuration

Regular IDC tree  
Send calling party DID disabled

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

	Incoming Digits *	Converted Digits	CPND Name	CPND language
1	647XXX1257	1257	.	Roman characters
2	647XXX1258	1258	.	Roman characters
3	647XXX1259	1259	.	Roman characters
4	647XXX1260	1260	.	Roman characters
5	647XXX1264	1700	.	Roman characters

**Figure 47 – Digit Conversion Tree**

### 5.6.6. Outbound Call - Special Number Configuration

There are special numbers which have been configured to be used for this testing such as: 0, 1800, 411, 911 and so on.

Select **Dialing and Numbering Plans** → **Electronic Switched Network** from the left pane to display the **Electronic Switched Network (ESN)** screen as show in **Figure 39**. Select **Special Number (SPN)**. Enter a SPN number and then click on the **to Add** button. **Figure 48** shows all the special numbers used for this testing.

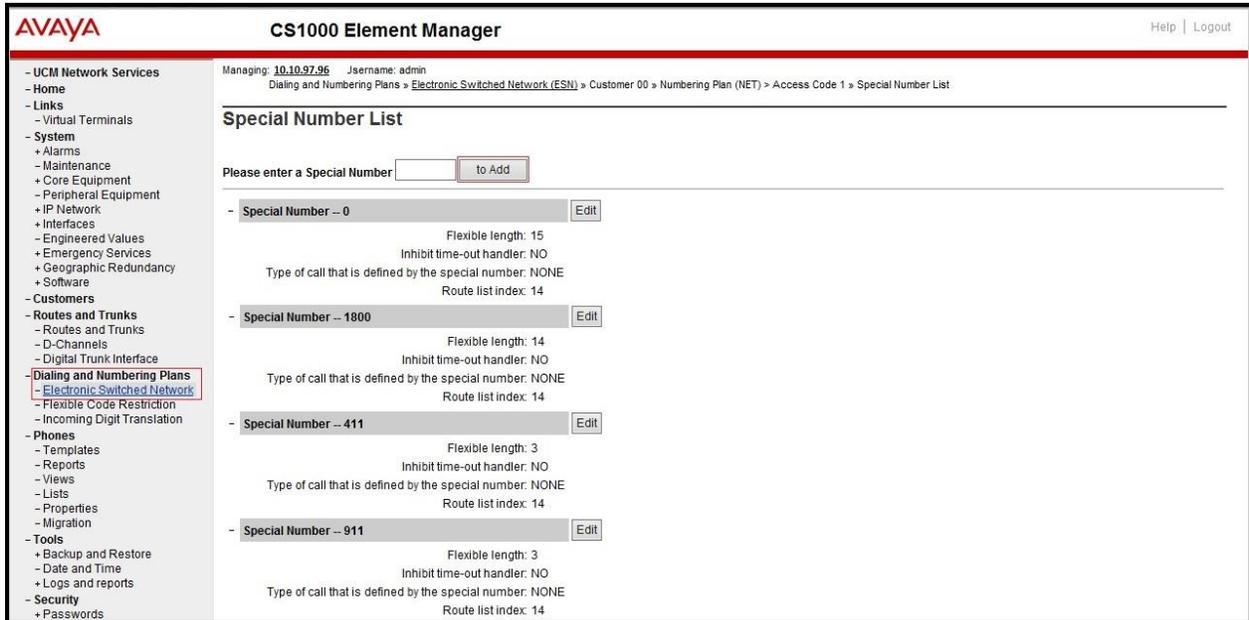
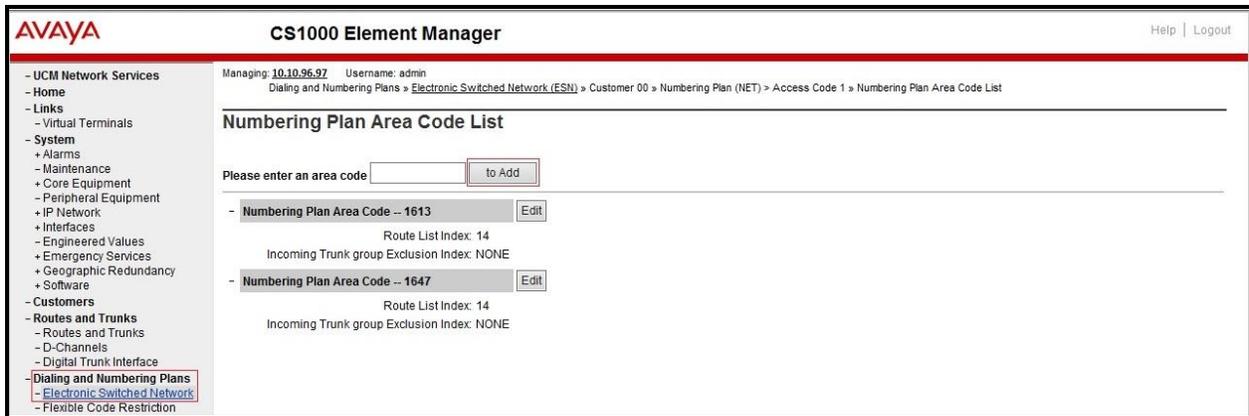


Figure 48 – SPN numbers

### 5.6.7. 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 (not shown). Select **Numbering Plan Area Code (NPA)** as shown in **Figure 39**. Enter the area code desired in the textbox and click on the **to Add** button. The 1613, and 1647 area codes were used in this configuration as shown in **Figure 49**.



**Figure 49 – Numbering Plan Area List**

## 5.7. Administer a Phone

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

### 5.7.1. Phone creation

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

```
>ld 11
REQ: new
TYPE: 2002p2
TN 96 0 0 2
DATE
PAGE
DES
MODEL_NAME
EMULATED
DES 2002P2 ← Describe information for IP Phone
TN 96 0 00 02 VIRTUAL ← Set Terminal Number for IP Phone
TYPE 2002P2
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 00010 ← Set bandwidth zone for IP phone
CUR_ZONE 00010
MRT
ERL 12345
ECL 0
FDN
TGAR 0
LDN NO
NCOS 7
```

```

SGRP 0
RNPG 0
SCI 0
SSU
LNRS 16
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS UNR FBA WTA LPR MTD FNA HTA TDD CRPD
  MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
  POD SLKD CCSD SWD LNA CNDA
  CFTD SFA MRD DDV CNIA CDCA MSID DAPA BFED RCBD
  ICDD CDMD LLCN MCTD CLBD AUTU
  GPUD DPUD DNDD CFXA ARHD CLTD ASCD
  CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
  UDI RCC HBTD AHD IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
  DRDD EXR0
  USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
  FSDS 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 1257 0  MARP ← Set the position of DN 1257 to display on key 0 of the phone
  CPND
  CPND_LANG ROMAN
  NAME MTS_01 ← Set name to display
  XPLN 13
  DISPLAY_FMT FIRST, LAST
  01
<Text removed for brevity>

```

### 5.7.2. Enable Privacy for the Phone

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

To hide the display number, set **CLS** (Class of Service) to **DDGD**. CS1000 will include “Privacy:id” in the SIP message header before sending it to MTS Allstream SIP Trunk Service.

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

To allow the display number, set **CLS** to **DDGA**. CS1000 will not send the Privacy header to MTS Allstream SIP Trunk Service.

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

### 5.7.3. Enable Call Forward for Phone

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

Select **Customer** → **00** → **Call Redirection**. The Call Redirection page is shown in **Figure 50**.

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

The screenshot displays the 'Call Redirection' configuration page in the AVAYA CS1000 Element Manager. The page is divided into several sections:

- Call redirection by day:** Includes four input fields for 'Days for day option 0' through 'Days for day option 3'.
- Redirection Holidays:** Includes a checkbox for 'Do not disturb hunting' and a dropdown menu for 'Total redirection count limit' set to 0.
- Options:** Includes checkboxes for 'Call forward reminder tone for 500/2500 sets', 'CFNA treatment for call waiting calls on a DN', 'DID call to second degree busy treatment', 'Message center' (checked), and 'Prevention of reciprocal call forward' (checked).
- Call forward:** Includes radio buttons for 'Originating' (selected) and 'Forwarding'.
- Number of normal ringing cycles for CFNA:** Includes three dropdown menus for 'Option 0', 'Option 1', and 'Option 2', all set to 3.
- Number of distinctive ringing cycles for CFNA:** Includes three dropdown menus for 'Option 0', 'Option 1', and 'Option 2', all set to 3.
- Calls routed to message center:** Includes checkboxes for 'No answer DID calls', 'No answer non-DID calls', and 'DID calls to busy telephones'.

At the bottom right of the page, there are 'Save' and 'Cancel' buttons.

**Figure 50 – Call Redirection**

To enable Call Forward All Call (CFAC) feature for a phone over SIP trunk, use **ld 11**. Change its **CLS** to **CFXA**, and **SFA**, then program the forward number on the phone set. The following is the configuration of a phone that has CFAC enabled with forwarding number **61613XXX5206**.

```
>ld 11
REQ: chg
TYPE: 2002P2
TN 96 0 0 2

ECHG yes
ITEM CLS CFXA SFA
ITEM key 19 CFW 16 61613XXX5206
```

To enable Call Forward Busy (CFB) feature for phone over SIP trunk, use **ld 11**. Change its **CLS** to **FBA**, **HTA**, and **SFA**, then program the forward number as **HUNT** and **FDN**. The following is the configuration of a phone with CFB enabled to forwarding number **61613XXX5206**.

```
>ld 11
REQ: chg
TYPE: 2002P2
TN 96 0 0 2
ECHG yes
ITEM CLS FBA HTA SFA
ITEM HUNT 61613XXX5206
ITEM FDN 61613XXX5206
```

To enable Call Forward No Answer (CFNA) feature for a phone over SIP trunk, use **ld 11**. Change its **CLS** to **FNA**, and **SFA**, then program the forward number as **HUNT** and **FDN**. The following is the configuration of a phone that has CFNA enabled with forwarding number **61613XXX5206**.

```
>ld 11
REQ: chg
TYPE: 2002P2
TN 96 0 0 2
ECHG yes
ITEM CLS FNA SFA
ITEM HUNT 61613XXX5206
ITEM FDN 61613XXX5206
```

## 6. Configure Avaya Aura<sup>®</sup> Session Manager

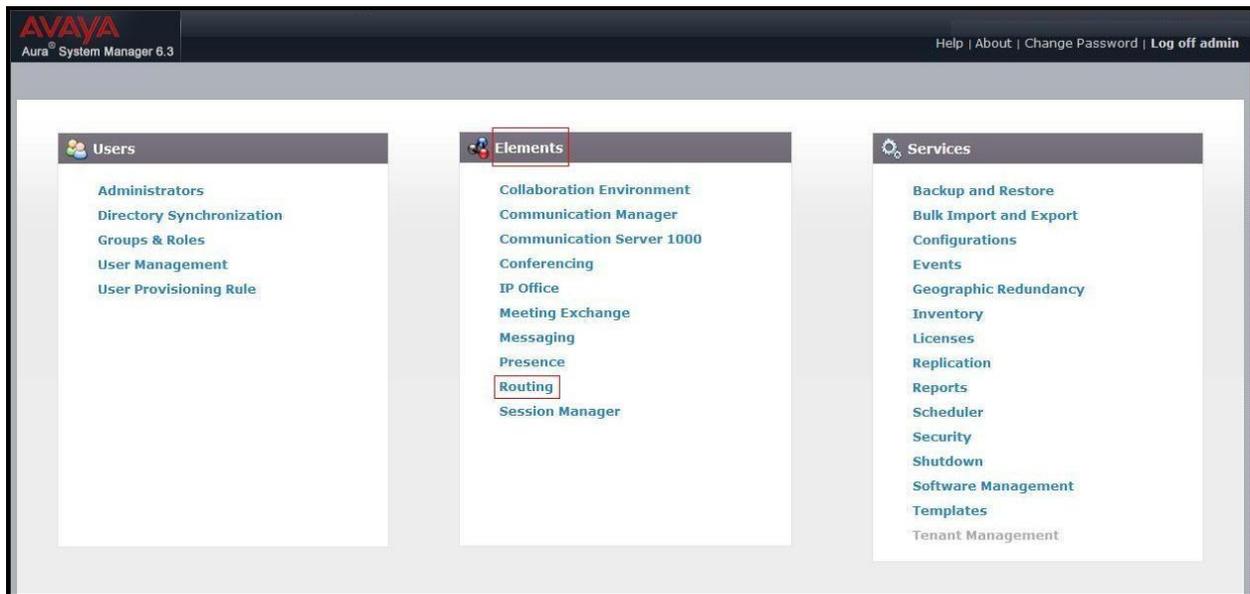
This section provides the procedures for configuring Session Manager. The procedures include configuring the following items:

- SIP Domain.
- Logical/physical Location that can be occupied by SIP Entities.
- Adaptation module to perform dial plan manipulation.
- SIP Entities corresponding to CS1000, Avaya SBCE and Session Manager.
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities.
- Routing Policies, which define route destinations and control call routing between the SIP Entities.
- Dial Patterns, which specify dialed digits and govern which Routing Policy is used to service a call.

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

## 6.1. Avaya Aura® System Manager Login and Navigation

Session Manager configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL “https://<ip-address>/SMGR”, where “<ip-address>” is the IP address of System Manager. At the **System Manager Log On** screen, enter an appropriate **User ID** and **Password** and press the **Log On** button (not shown). The initial screen shown below is then displayed.



**Figure 51 – System Manager Home Screen**

Most of the configuration items are performed in the Routing Element. Click on **Routing** in the **Elements** column to bring up the **Introduction to Network Routing Policy** screen.

The navigation tree displayed in the left pane will be referenced in subsequent sections to navigate to items requiring configuration.

The screenshot shows the Avaya Aura System Manager 6.3 interface. The top header includes the Avaya logo, 'Aura System Manager 6.3', and navigation links for 'Help | About | Change Password | Log off admin'. The left navigation pane is expanded to show the 'Routing' section, which includes sub-items: Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area displays the 'Introduction to Network Routing Policy' page, which explains that Network Routing Policy consists of several routing applications like 'Domains', 'Locations', 'SIP Entities', etc. It provides a recommended order for configuration:

**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"

**Figure 52 – Network Routing Policy**

## 6.2. Specify SIP Domain

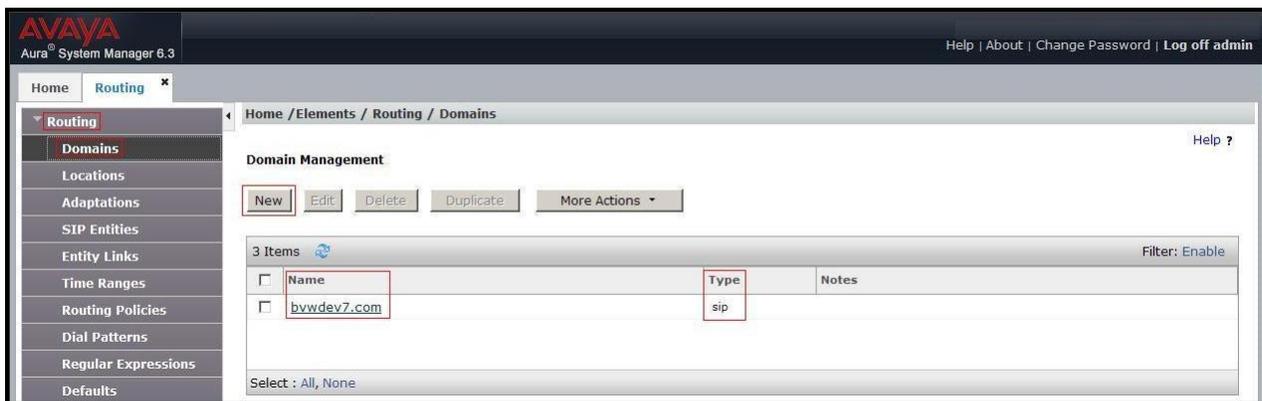
Create a SIP Domain for each domain of which Session Manager will need to be aware in order to route calls. For the compliance test, this includes the enterprise domain **bvwdev7.com**.

Navigate to **Routing → Domains** in the left-hand navigation pane and click the **New** button in the right pane. In the new right pane that appears (not shown), fill in the following:

- **Name:** Enter the domain name (refer to **Section 5.5.2**).
- **Type:** Select **sip** from the pull-down menu.
- **Notes:** Add a brief description (optional).

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

The screen below shows the existing entry for the enterprise domain.



**Figure 53 – Domain Management**

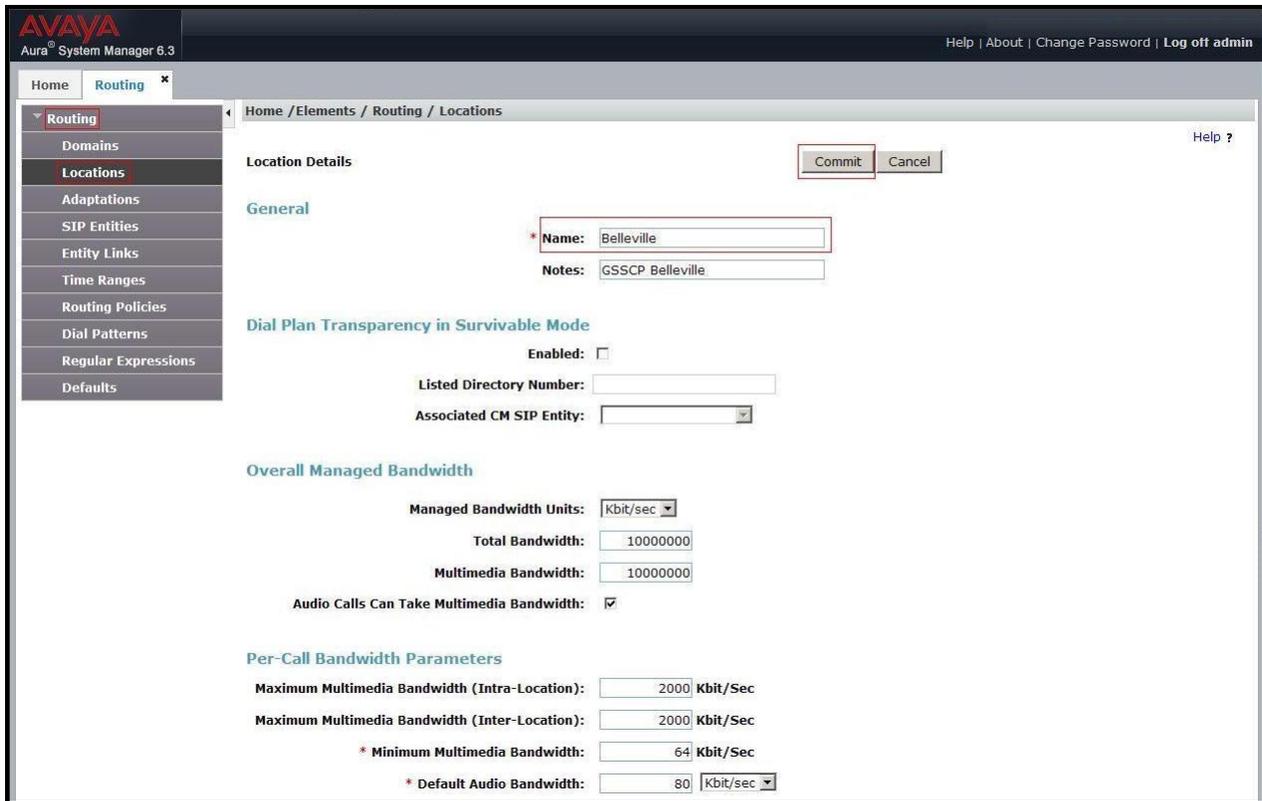
## 6.3. Add Location

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management and call admission control. A single Location was defined for the enterprise even though multiple subnets were used. The screens below show the addition of the Location named **Belleville**, which includes all equipment in the enterprise including CS1000, Session Manager and Avaya SBCE.

To add a Location, navigate to **Routing → Locations** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

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

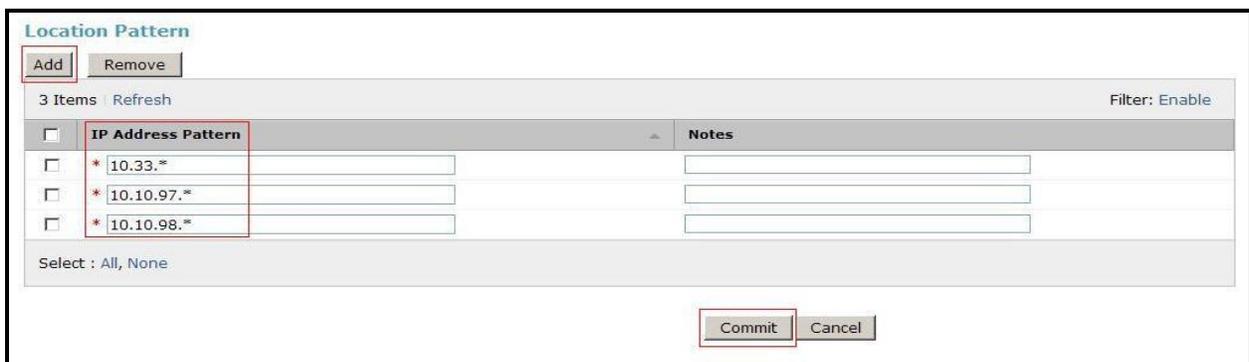
- **Name:** Enter a descriptive name for the Location.
- **Notes:** Add a brief description (optional).



**Figure 54 – Location Configuration**

In the **Location Pattern** section, click **Add** to enter IP Address patterns. The following patterns were used in testing:

- **IP Address Pattern:** 10.33.\*, 10.10.97.\*, 10.10.98.\*



**Figure 55 – IP Ranges Configuration**

Click **Commit** to save.

Note that call bandwidth management parameters should be set per customer requirement.

## 6.4. Configure Adaptations

An Adaptation is configured to format the History Info on CS1000 to be compatible with other Avaya products. To add a new adaptation, select **Routing** → **Adaptations**. Click the **New** button in the right pane (not shown). Enter an appropriate **Adaptation Name** to identify the adaptation. Select **CS1000Adapter** from the **Module Name** drop-down menu. Select **Name-Value Parameter** from the **Module Parameter Type** drop-down menu. Click **Add** button to add **Name** as **fromto** and **Value** as **true**. Click the **Commit** button after changes are completed.

The screenshot shows the Avaya Aura System Manager 6.3 interface. The left sidebar contains a navigation menu with 'Routing' selected. The main content area is titled 'Home / Elements / Routing / Adaptations'. The 'Adaptation Details' section is active, showing the 'General' tab. The 'Adaptation Name' field contains 'CS1K76\_Adaptation'. The 'Module Name' dropdown is set to 'CS1000Adapter'. The 'Module Parameter Type' dropdown is set to 'Name-Value Parameter'. Below these fields are 'Add' and 'Remove' buttons. A table with two columns, 'Name' and 'Value', contains one row with 'fromto' and 'true'. Below the table is a 'Select : All, None' dropdown. At the bottom, there are fields for 'Egress URI Parameters' and 'Notes'. The 'Commit' and 'Cancel' buttons are located at the top right of the form area.

Figure 56 - CS1000 Adaptation

An Adaptation is configured to convert the History Info to Diversion Header and to remove MIME. To add a new adaptation, select **Routing** → **Adaptations**. Click the **New** button in the right pane (not shown). Enter an appropriate **Adaptation Name** to identify the adaptation. Select **DiversionTypeAdapter** from the **Module Name** drop-down menu. Select **Name-Value Parameter** from the **Module Parameter Type** drop-down menu. Click **Add** button to add **Name** as **MIME** and **Value** as **no**. Click the **Commit** button after changes are completed.

The screenshot shows the Avaya Aura System Manager 6.3 interface. The left sidebar contains a navigation menu with 'Routing' selected. The main content area is titled 'Home / Elements / Routing / Adaptations'. The 'Adaptation Details' section is active, showing the 'General' tab. The 'Adaptation Name' field contains 'Diversion-Type-Remove-MIME'. The 'Module Name' dropdown is set to 'DiversionTypeAdapter'. The 'Module Parameter Type' dropdown is set to 'Name-Value Parameter'. Below these fields are 'Add' and 'Remove' buttons. A table with two columns, 'Name' and 'Value', contains one row with 'MIME' and 'no'. Below the table is a 'Select : All, None' dropdown. At the bottom, there are fields for 'Egress URI Parameters' and 'Notes'. The 'Commit' and 'Cancel' buttons are located at the top right of the form area.

Figure 57 – Diversion Header Adaptation

## 6.5. Add SIP Entities

A SIP Entity must be added for Session Manager and for each SIP telephony system connected to Session Manager which includes CS1000 and Avaya SBCE. Navigate to **Routing → SIP Entities** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

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

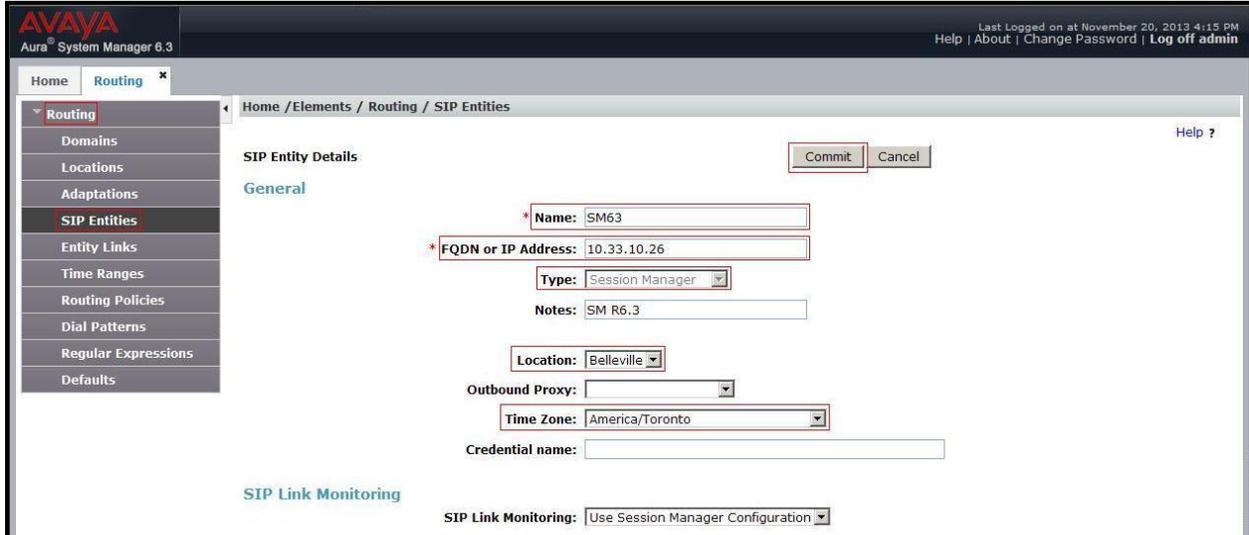
- **Name:** Enter a descriptive name.
- **FQDN or IP Address:** Enter the FQDN or IP address of the SIP Entity that is used for SIP signaling.
- **Type:** Select **Session Manager** for Session Manager, **Other** for CS1000 and Avaya SBCE.
- **Adaptation:** This field is only present if **Type** is not set to **Session Manager**. If applicable, select the appropriate Adaptation module that will be applied to the SIP Entity being created.
- **Location:** Select the Location that applies to the SIP Entity being created. For the compliance test, all components were located in Location **Belleville**.
- **Time Zone:** Select the time zone for the Location above.

In this configuration, there are three SIP Entities.

- Session Manager SIP Entity
- Communication Manager 1000 SIP Entity
- Avaya Session Border Controller for Enterprise SIP Entity

### 6.5.1. Configure Session Manager SIP Entity

The following screen shows the addition of the Session Manager SIP Entity named **SM63**. The IP address of Session Manager's signaling interface **10.33.10.26** is entered for **FQDN or IP Address**. The **Location** field is set to **Belleville**. Select **Time Zone** as **America/Toronto**.



**Figure 58 – Session Manager SIP Entity**

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

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

- **Port:** Port number on which Session Manager listens for SIP requests.
- **Protocol:** Transport protocol to be used with this port.
- **Default Domain:** The default domain associated with this port. For the compliance test, this was the enterprise SIP Domain.

Defaults can be used for the remaining fields. Click the **Commit** button (not shown) to save.

The compliance test used port **5060** with **UDP** for connecting to CS1000 and Avaya SBCE.

**Port**

TCP Failover port:

TLS Failover port:

2 Items | Refresh Filter: Enable

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	5060	UDP	bvwdev7.com	<input type="text"/>

Select : All, None

**Figure 59 – Session Manager SIP Entity Port**

### 6.5.2. Configure Communication Server 1000 SIP Entity

The following screen shows the addition of the CS1000 SIP Entity named **car3-ssg-carrier**. In order for Session Manager to send SIP service provider traffic on a separate Entity Link to CS1000, it is necessary to create a separate SIP Entity for CS1000, in addition to the one created at Session Manager installation, for use with all other SIP traffic within the enterprise. The **FQDN or IP Address** field is set to the IP address of CS1000 signaling Node **10.10.97.178**. Select **Type** as **Other**. Select **Adaptation** as **CS1K76\_Adaptation** (created in **Section 6.4**). The **Location** field is set to **Belleville** which is the location that includes the subnet where CS1000 resides. Select **Time Zone** as **America/Toronto**.

**AVAYA**  
Aura System Manager 6.3 Last Logged on at October 21, 2014 4:04 AM

Home Routing

Home / Elements / Routing / SIP Entities Help ?

**SIP Entity Details**

**General**

\* Name:

\* FQDN or IP Address:

Type:

Notes:

Adaptation:

Location:

Time Zone:

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

Credential name:

Call Detail Recording:

CommProfile Type Preference:

**Figure 60 – Communication Server 1000 SIP Entity**

### 6.5.3. Configure Avaya SBCE SIP Entity

The following screen shows the SIP Entity for the Avaya SBCE. The **FQDN or IP Address** field is set to the IP address of the Avaya SBCE's private network interfaces. Select **Type** as **Other**. Select **Adaptation** as **Diversion-Type-Remove-MIME** (created in **Section 6.4**). The **Location** field is set to **Belleville** which includes the subnet where Avaya SBCE resides. Select **Time Zone** as **America/Toronto**.

The following screenshot shows the SIP Entity for Avaya SBCE.

The screenshot displays the Avaya Aura System Manager 6.3 web interface. The breadcrumb navigation shows 'Home / Elements / Routing / SIP Entities'. The left-hand navigation menu is expanded to 'SIP Entities'. The main content area is titled 'SIP Entity Details' and includes 'Commit' and 'Cancel' buttons. Under the 'General' tab, the following fields are visible:

- Name:** SBCE
- FQDN or IP Address:** 10.10.98.13
- Type:** Other (dropdown menu)
- Notes:** (empty text area)
- Adaptation:** Diversion-Type-Remove-MIME (dropdown menu)
- Location:** Belleville (dropdown menu)
- Time Zone:** America/Toronto (dropdown menu)
- SIP Timer B/F (in seconds):** 4
- Credential name:** (empty text field)
- Call Detail Recording:** none (dropdown menu)
- CommProfile Type Preference:** (empty dropdown menu)

Figure 61 – Avaya SBCE SIP Entity

## 6.6. Add Entity Links

A SIP trunk between Session Manager and a telephony system is described by an Entity Link. Two Entity Links were created: one to CS1000 and one to Avaya SBCE.

To add an Entity Link, navigate to **Routing → Entity Links** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Name:** Enter a descriptive name.
- **SIP Entity 1:** Select the Session Manager being used.
- **Protocol:** Select the transport protocol used for this link.
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end.
- **SIP Entity 2:** Select the name of the other system as defined in **Section 6.5**.
- **Port:** Port number on which the other system receives SIP requests from the Session Manager.
- **Connection Policy:** Select **trusted**. Note: If this box is not selected as trusted, calls from the associated SIP Entity specified in **Section 6.5** will be denied.

Click **Commit** to save.

The following screen illustrates the Entity Link to the CS1000. The protocol and ports defined here must match the values used for the CS1000 signaling in **Section 5.5.2**.

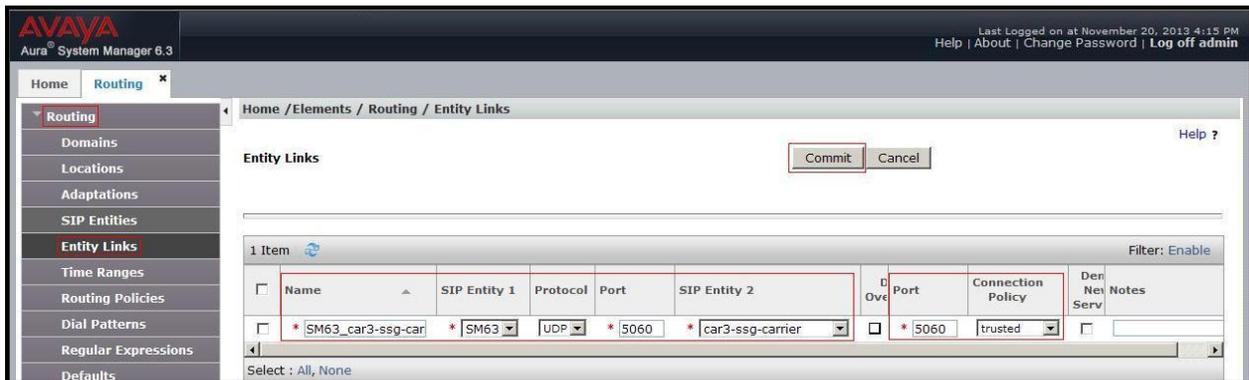


Figure 62 – Communication Server 1000 Entity Link

The following screen illustrates the Entity Link to Avaya SBCE. The protocol and ports defined here must match the values used for Avaya SBCE mentioned in **Section 7.2.4**, later in this document.



**Figure 63 – Avaya SBCE Entity Link**

## 6.7. Configure Time Ranges

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

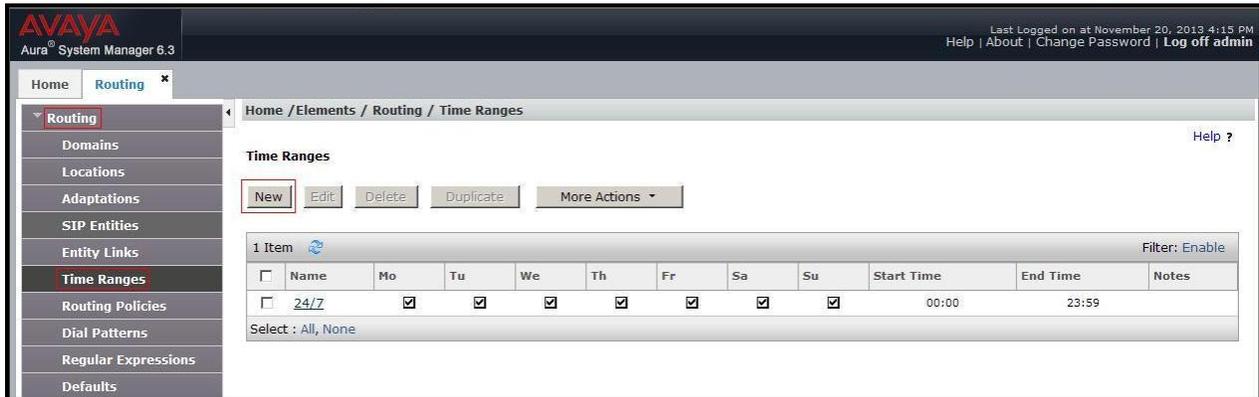


Figure 64 – Time Ranges

## 6.8. Add Routing Policies

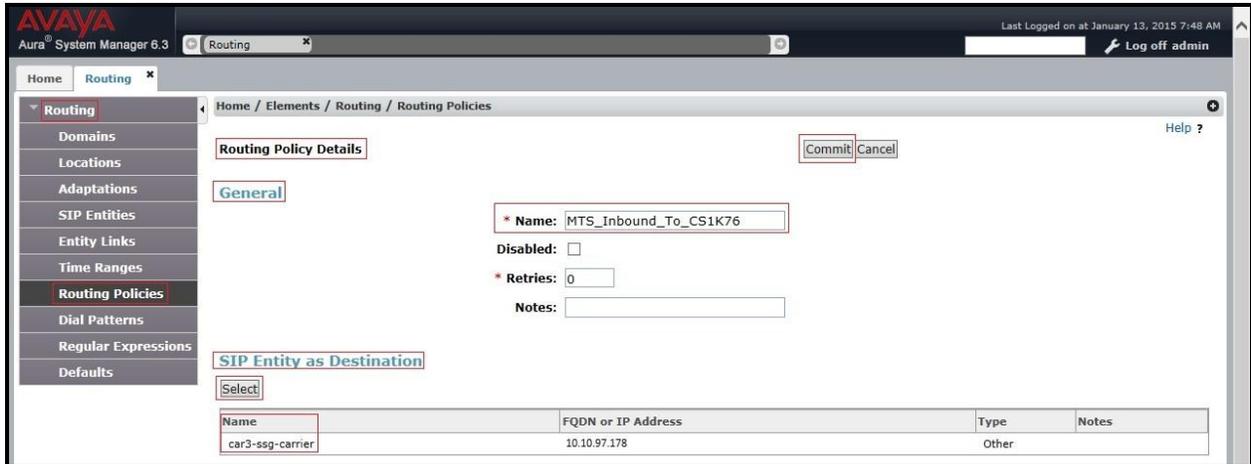
Routing Policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.5**. Two Routing Policies must be added: one for the CS1000 and one for Avaya SBCE. To add a Routing Policy, navigate to **Routing** → **Routing Policies** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown on the next page), fill in the following:

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

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

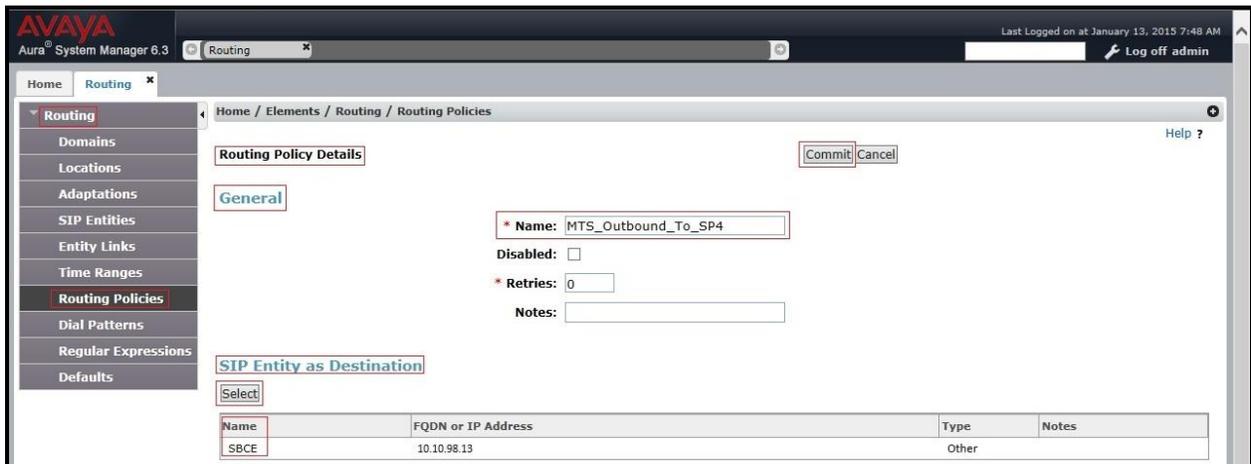
In the **SIP Entity as Destination** section, click **Select**. The **SIP Entity List** page opens (not shown). Select the appropriate SIP Entity to which this Routing Policy applies and click **Select**. The selected SIP Entity displays on the **Routing Policy Details** page as shown below. Use default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

The following screen shows the **Routing Policy Details** for the policy named **MTS\_Inbound\_To\_CS1K76** associated with incoming PSTN calls from MTS Allstream SIP Trunk Service to the CS1000. Observe the **SIP Entity as Destination** is the entity named **car3-ssg-carrier**.



**Figure 65 – Routing to Communication Server 1000**

The following screen shows the **Routing Policy Details** for the policy named **MTS\_Outbound\_To\_SP4**. This is associated with outgoing calls from the CS1000 to the PSTN via MTS Allstream SIP Trunk Service, through Avaya SBCE. Observe the **SIP Entity as Destination** is the entity named **SBCE**.



**Figure 66 – Routing to Avaya SBCE**

## 6.9. Add Dial Patterns

Dial Patterns are used to route calls through Session Manager. For the compliance test, Dial Patterns were configured to route calls from the CS1000 to MTS Allstream SIP Trunk Service and vice versa. Dial Patterns define which Route Policy will be selected as route destination for a particular call based on the dialed digits, destination Domain and originating Location.

To add a Dial Pattern, navigate to **Routing → Dial Patterns** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

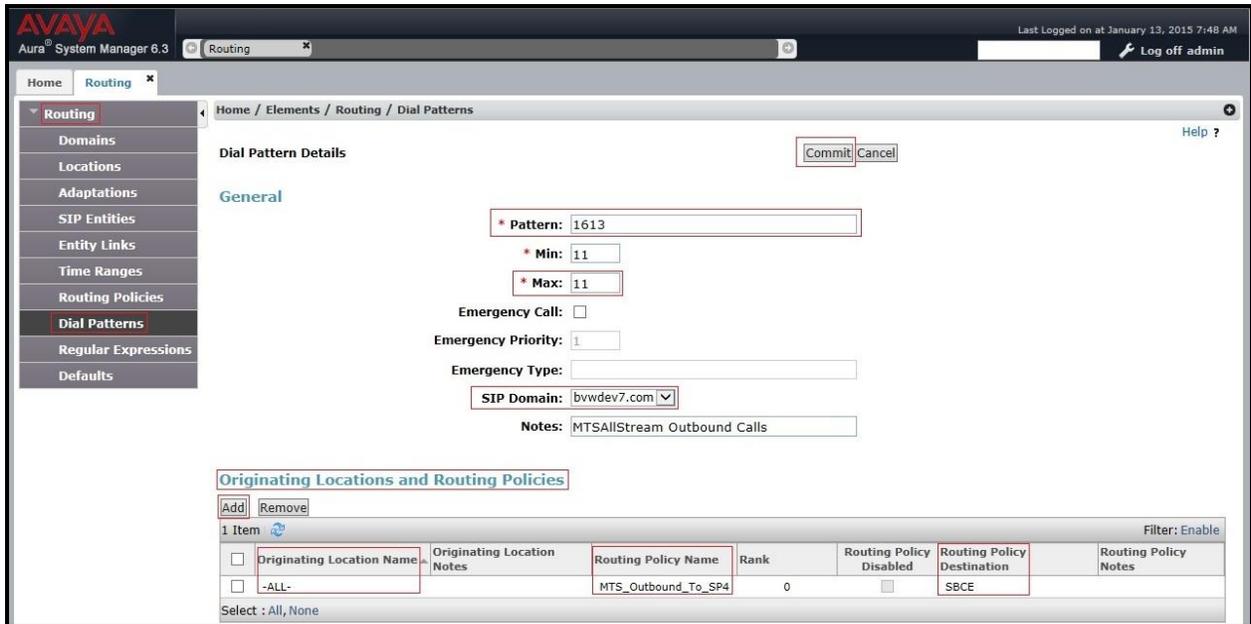
In the **General** section, enter the following values. Use default values for all remaining fields.

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

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

Default values can be used for the remaining fields. Click **Commit** to save. Two examples of the Dial Patterns used for the compliance test are shown below, one for outbound calls from the enterprise to the PSTN and one for inbound calls from the PSTN to the enterprise. Other Dial Patterns (e.g., 1800, 911, etc.) were similarly defined.

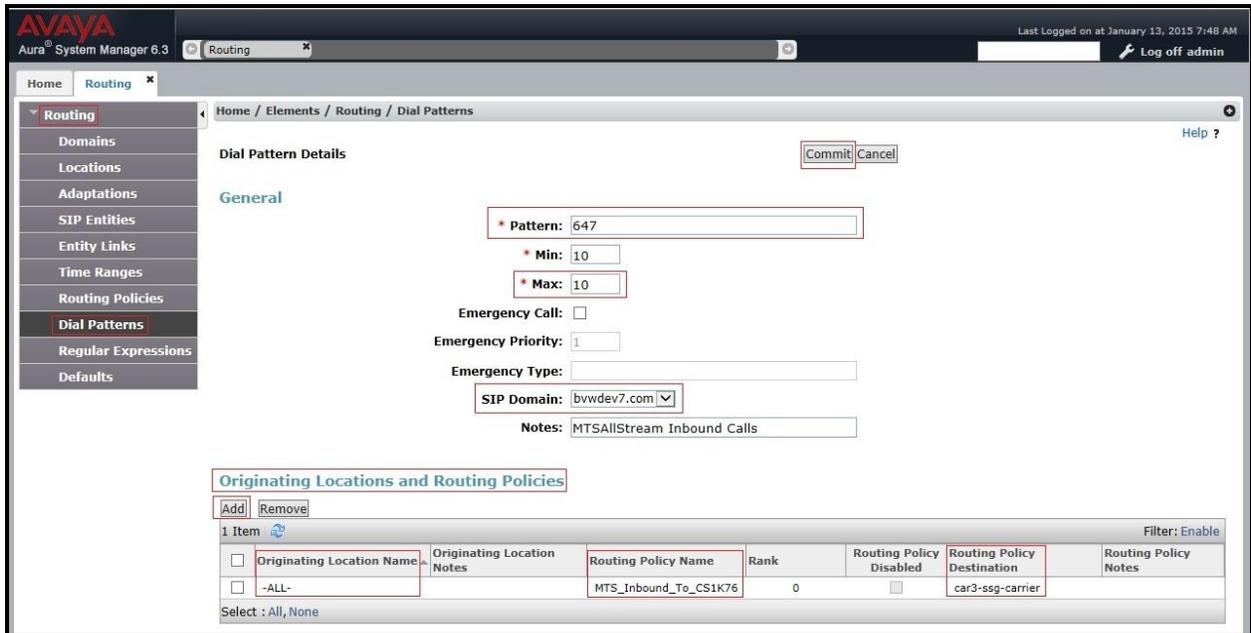
The following screen shows that outbound dialed numbers with a maximum of 11 digits that begin with **1613** and have a destination SIP Domain of **bvwddev7.com** use the Routing Policy Names **MTS\_Outbound\_To\_SP4** as defined in **Section 6.8**.



**Figure 67 – Dial Pattern 1613**

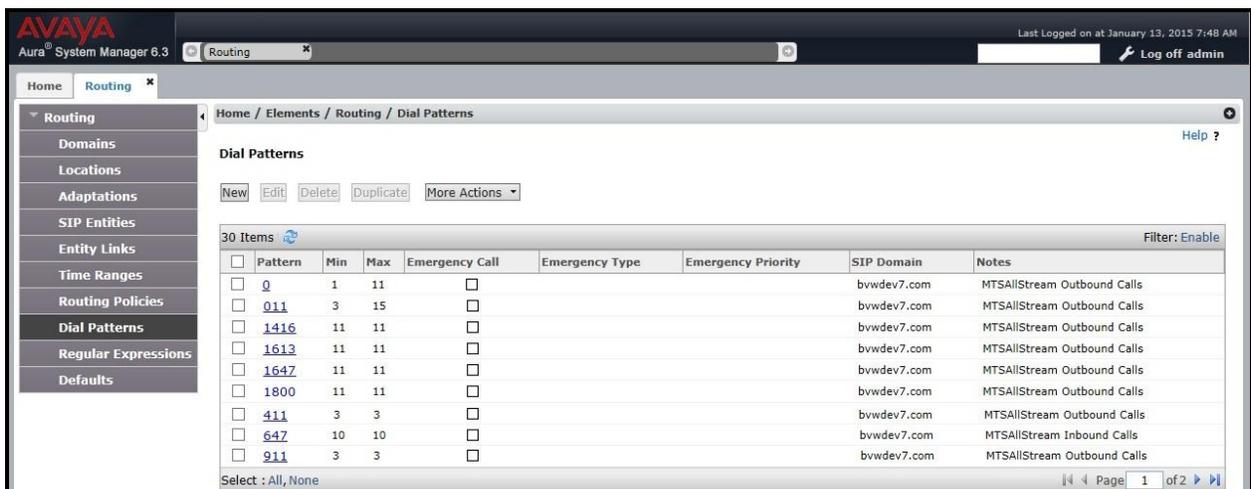
Note that the above Dial Pattern did not restrict outbound calls to specific US/Canada area codes. In real deployments, appropriate restriction can be exercised per customer business policies.

The following screen shows that inbound 10-digit numbers that start with **647** use Routing Policy Name **MTS\_Inbound\_To\_CS1K76** as defined in **Section 6.8**. This Dial Pattern matches the DID numbers assigned to the enterprise by MTS Allstream SIP Trunk Service.



**Figure 68 – Dial Pattern 647**

The following screen illustrates a list of dial patterns used for inbound and outbound calls between the enterprise and the PSTN.



**Figure 69 – Dial Pattern List**

## 7. Configure Avaya Session Border Controller for Enterprise

This section describes the configuration of Avaya SBCE necessary for interoperability with the Session Manager and MTS Allstream SIP Trunk Service.

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

**Note:** The following section assumes that Avaya SBCE has been installed and that network connectivity exists between the systems. For more information on Avaya SBCE, see relevant product documentation references in **Section 11** of these Application Notes.

### 7.1. Log into the SBCE

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

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



**AVAYA**

**Session Border Controller  
for Enterprise**

**Log In**

Username:

Password:

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

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

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

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**Figure 70 – Avaya SBCE Login**

## 7.2. Global Profiles

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

### 7.2.1. Configure Server Interworking - Avaya Site

Server Interworking allows 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**

- Select **avaya-ru** in Interworking Profiles.
- Click **Clone**.
- Enter **Clone Name: SM63** and click **Finish** (not shown).

From the list of **Interworking Profiles**, click on **SM63** to edit.

- On the **General** tab, set **T.38 Support** as **Yes** (if using Fax T.38) or **No** (if using Fax G.711 pass-through). Other options can be left at default.
- On the **Timers**, **URI Manipulation**, **Header Manipulation** and **Advanced** tabs, all options can be left at default. Click **Finish** (not shown).

The following screen shows that Session Manager server interworking profile (named: **SM63**) was added.

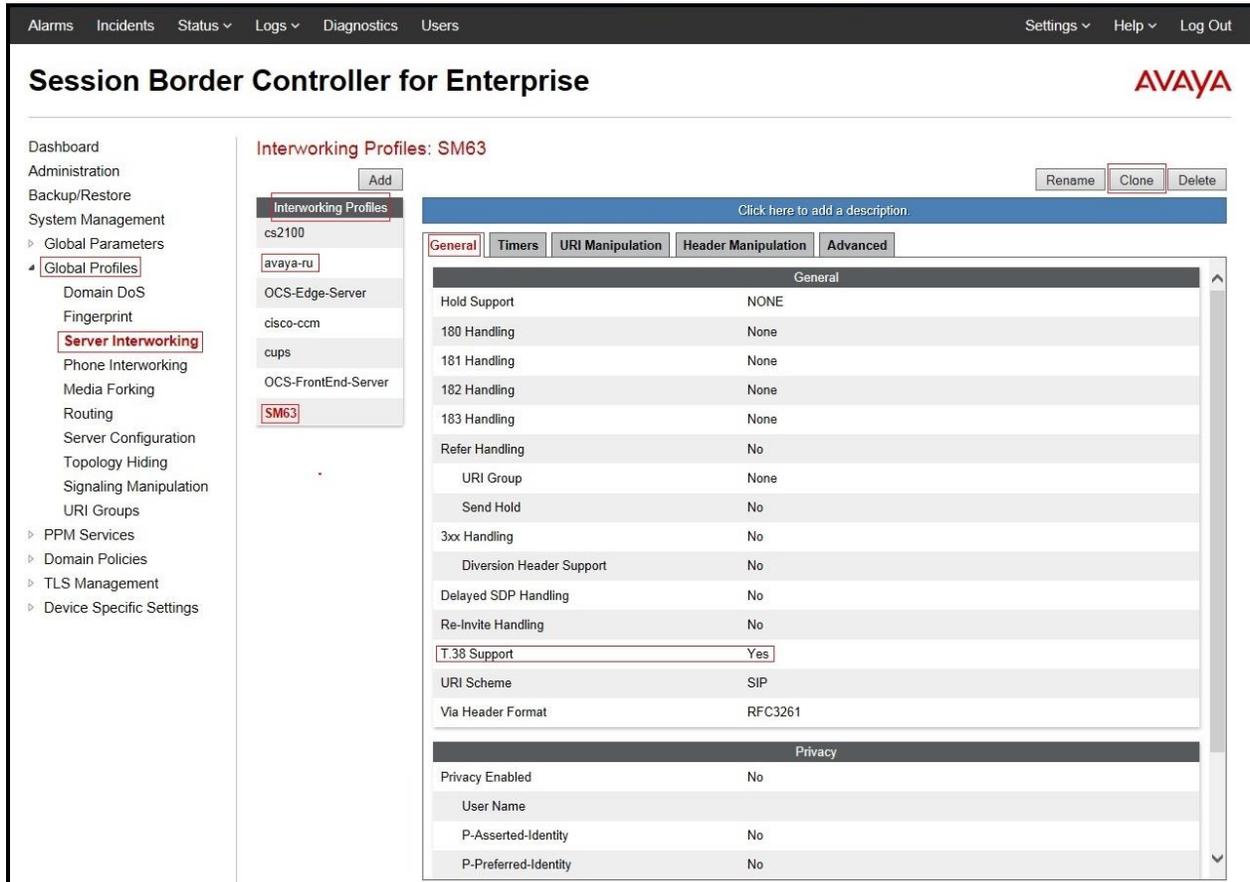


Figure 71 - Server Interworking – Avaya site

## 7.2.2. Configure Server Interworking – MTS Allstream Site

From the menu on the left-hand side, select **Global Profiles** → **Server Interworking** and click **Add** as highlighted below.

- Enter **Profile Name: SP4**.
- On the **General** tab, set **T.38 Support** as **Yes** (if using Fax T.38) or **No** (if using Fax G.711 pass-through). Other options can be left at default.
- On the **Timers**, **URI Manipulation**, **Header Manipulation** and **Advanced** tabs, all options can be left at default. Click **Finish** (not shown).

The following screen shows that the MTS Allstream SIP Trunk Service interworking profile (named **SP4**) was added.

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

The left sidebar contains a navigation menu with categories like 'Dashboard', 'Administration', 'System Management', and 'Global Profiles'. Under 'Global Profiles', 'Server Interworking' is highlighted with a red box.

The main content area is titled 'Interworking Profiles: SP4'. It features an 'Add' button and 'Rename', 'Clone', and 'Delete' buttons. Below this is a list of interworking profiles, with 'SP4' selected and highlighted in red. The configuration for 'SP4' is shown in a table with tabs for 'General', 'Timers', 'URI Manipulation', 'Header Manipulation', and 'Advanced'. The 'General' tab is active, showing the following settings:

General	
Hold Support	NONE
180 Handling	None
181 Handling	None
182 Handling	None
183 Handling	None
Refer Handling	No
URI Group	None
3xx Handling	No
Diversion Header Support	No
Delayed SDP Handling	No
Re-Invite Handling	No
<b>T.38 Support</b>	<b>Yes</b>
URI Scheme	SIP
Via Header Format	RFC3261
Privacy	
Privacy Enabled	No
User Name	
P-Asserted-Identity	No
P-Preferred-Identity	No
Privacy Header	

**Figure 72 - Server Interworking – MTS Allstream site**

### 7.2.3. Configure URI Groups

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

The following URI Group configuration is used for the compliance test in a lab environment where equipment is for shared use. The URI-Group named **SP4** was used to match the “From” and “To” headers in a SIP call dialog received from both Enterprise and MTS Allstream SIP Trunk Service. If there is a match, the Avaya SBCE will apply the appropriate Routing Profiles (see **Section 7.2.6, 7.2.7**), and Server Flows (see **Section 7.4.4**) to route incoming and outgoing calls to the right destinations. In the production environment, there is not a requirement to define this URI Group.

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

- Enter **Group Name: SP4**.
- Edit the **URI Type: Regular Expression** (not shown).
- **Add URI: .\*10\10\98\111** (Avaya SBCE public interface IP address), **.\*10\10\98\13** (Avaya SBCE internal interface IP address), **.\*192\168\2\12** (MTS Allstream Signaling Server IP address), **.\*192\168\2\13** (MTS Allstream Media Server IP address), **.\*anonymous\invalid** (Anonymous URI), **.\*bvwdev7\com** (Enterprise domain).
- Click **Finish** (not shown).

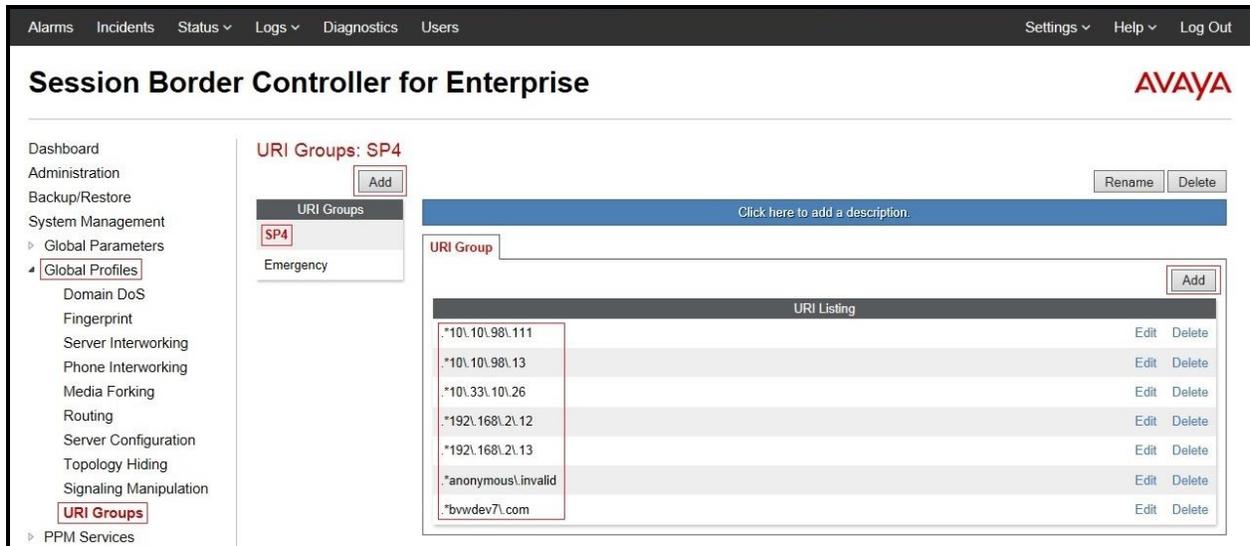


Figure 73 - URI Group

## 7.2.4. Configure Server – Session Manager

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

From the menu on the left-hand side, select **Global Profiles** → **Server Configuration** and click **Add** as highlighted below.

Enter **Profile Name: SM63**.

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

- **Server Type:** Select **Call Server**.
- **IP Address/FQDN:** **10.33.10.26** (Session Manager IP Address).
- **Port:** **5060**.
- **Transport:** **UDP**.
- Click **Finish** (not shown).

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes 'Alarms', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main header shows 'Session Border Controller for Enterprise' and the 'AVAYA' logo. A left-hand navigation menu lists various system management options, with 'Server Configuration' highlighted. The main content area is titled 'Server Configuration: SM63' and features an 'Add' button. Below this, there are tabs for 'General', 'Authentication', 'Heartbeat', and 'Advanced'. The 'General' tab is active, showing a 'Server Type' dropdown set to 'Call Server'. A table below lists the configuration details:

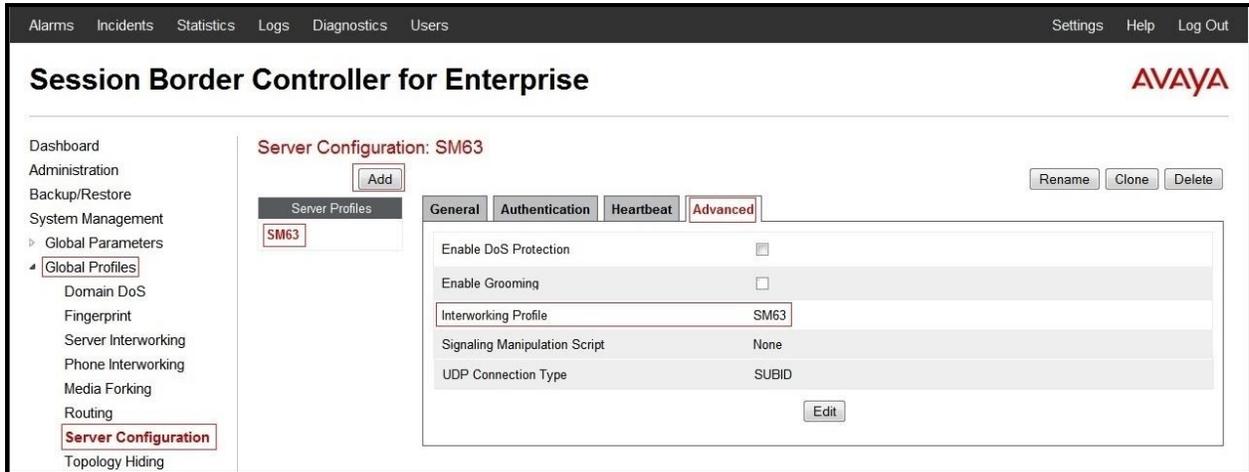
IP Address / FQDN	Port	Transport
10.33.10.26	5060	UDP

An 'Edit' button is located below the table. At the top right of the configuration area, there are 'Rename', 'Clone', and 'Delete' buttons.

**Figure 74 – Session Manager - General Server Configuration**

On the **Advanced** tab:

- Select **SM63** for **Interworking Profile** (Refer to **Section 7.2.1**).
- Click **Finish** (not shown).



**Figure 75 – Session Manager - Advanced Server Configuration**

## 7.2.5. Configure Server – MTS Allstream

From the menu on the left-hand side, select **Global Profiles** → **Server Configuration** and click **Add** as highlighted below.

Enter **Profile Name: SP4**.

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

- **Server Type:** Select **Trunk Server**.
- **IP Address/FQDN:** **192.168.2.12** (MTS Allstream Signaling Server IP Address).
- **Port:** **5060**.
- **Transport:** **UDP**.
- Click **Finish** (not shown).

The screenshot displays the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes 'Alarms', 'Incidents', 'Status', 'Logs', 'Diagnostics', 'Users', 'Settings', 'Help', and 'Log Out'. The main header shows 'Session Border Controller for Enterprise' and the 'AVAYA' logo. A left-hand navigation menu lists various system management options, with 'Server Configuration' highlighted. The main content area is titled 'Server Configuration: SP4' and features an 'Add' button. Below this, there are tabs for 'General', 'Authentication', 'Heartbeat', and 'Advanced'. The 'General' tab is active, showing a form with the following fields:

Server Type	IP Address / FQDN	Port	Transport
Trunk Server	192.168.2.12	5060	UDP

An 'Edit' button is located below the table. Additional buttons for 'Rename', 'Clone', and 'Delete' are visible in the top right corner of the configuration area.

**Figure 76 – MTS Allstream - General Server Configuration**

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

- **Interworking Profile:** Select **SP4** (Refer to **Section 7.2.2**).
- Click **Finish** (not shown).



**Figure 77 - MTS Allstream - Advanced Server Configuration**

### 7.2.6. Configure Routing – Avaya Site

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

From the menu on the left-hand side, select **Global Profiles** → **Routing** and click **Add** as highlighted below.

Enter **Profile Name:** **SP4\_To\_SM63** (not shown).

- **URI Group:** **SP4** (Refer to **Section 7.2.3**).
- **Load Balancing:** **Priority**.
- Check **Next Hop Priority**.
- Click **Add** button to add a Next-Hop Address
- **Priority/Weight:** **1**.
- **Server Configuration:** **SM63** (Refer to **Section 7.2.4**).
- **Next Hop Address:** **10.33.10.26:5060 (UDP)** (Session Manager IP address).
- Click **Finish**.

Alarms Incidents Status Logs Diagnostics Users Settings Help Log Out

## Session Border Controller for Enterprise

AVAYA

Dashboard  
Administration  
Backup/Restore  
System Management  
Global Parameters  
Global Profiles  
Domain DoS  
Fingerprint  
Server Interworking  
Phone Interworking  
Media Forking  
Routing  
Server Configuration  
Topology Hiding  
Signaling Manipulation  
URI Groups  
PPM Services  
Domain Policies  
TLS Management

Routing Profiles: SP4\_To\_SM64

Add

Rename Clone Delete

Click here to add a description.

Routing Profile

Routing Profile

URI Group SP4 Time of Day default

Load Balancing Priority NAPTR

Transport None Next Hop Priority

Next Hop In-Dialog Ignore Route Header

Add

Priority / Weight	Server Configuration	Next Hop Address	Transport
1	SM63	10.33.10.26:5060 (UDP)	None Delete

Back Finish

Transport  
UDP Edit Delete

**Figure 78 - Routing to Avaya**

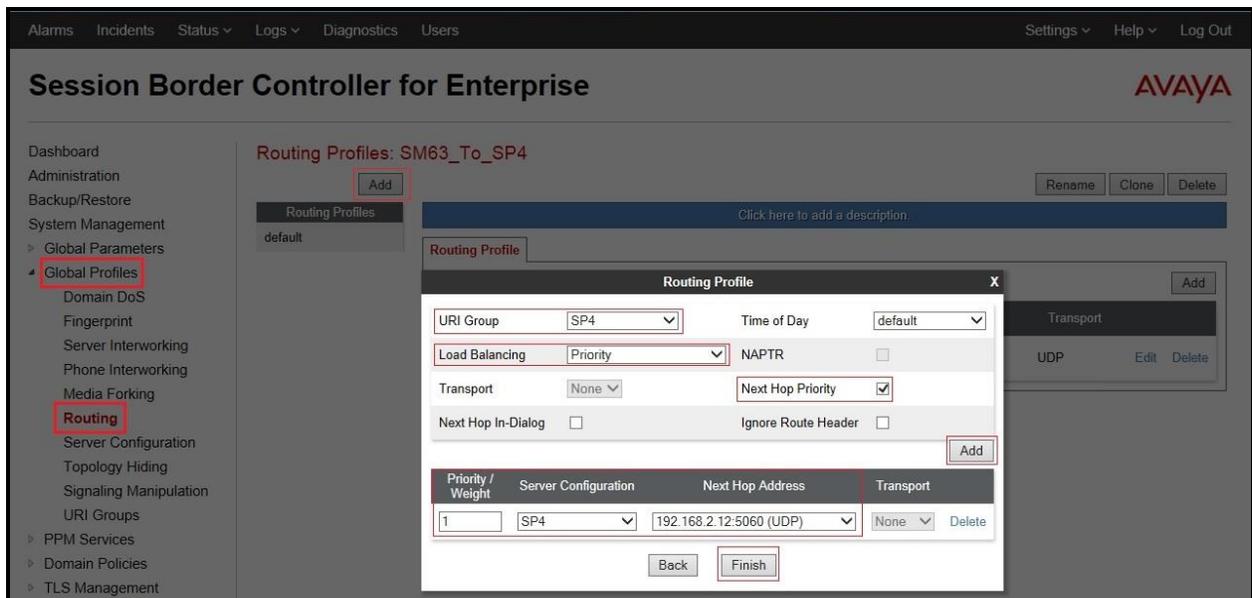
## 7.2.7. Configure Routing – MTS Allstream Site

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

From the menu on the left-hand side, select **Global Profiles** → **Routing** and click **Add** as highlighted below.

Enter **Profile Name: SM63\_To\_SP4** (not shown).

- **URI Group: SP4** (Refer to **Section 7.2.3**).
  - **Load Balancing: Priority**.
  - Check **Next Hop Priority**.
  - Click **Add** button to add a Next-Hop Address
  - **Priority/Weight: 1**.
  - **Server Configuration: SP4** (Refer to **Section 7.2.5**).
  - **Next Hop Address: 192.168.2.12:5060 (UDP)** (MTS Allstream Signaling Server IP address).
  - Click **Finish**.



**Figure 79 - Routing to MTS Allstream**

## 7.2.8. Configure Topology Hiding – Avaya Site

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

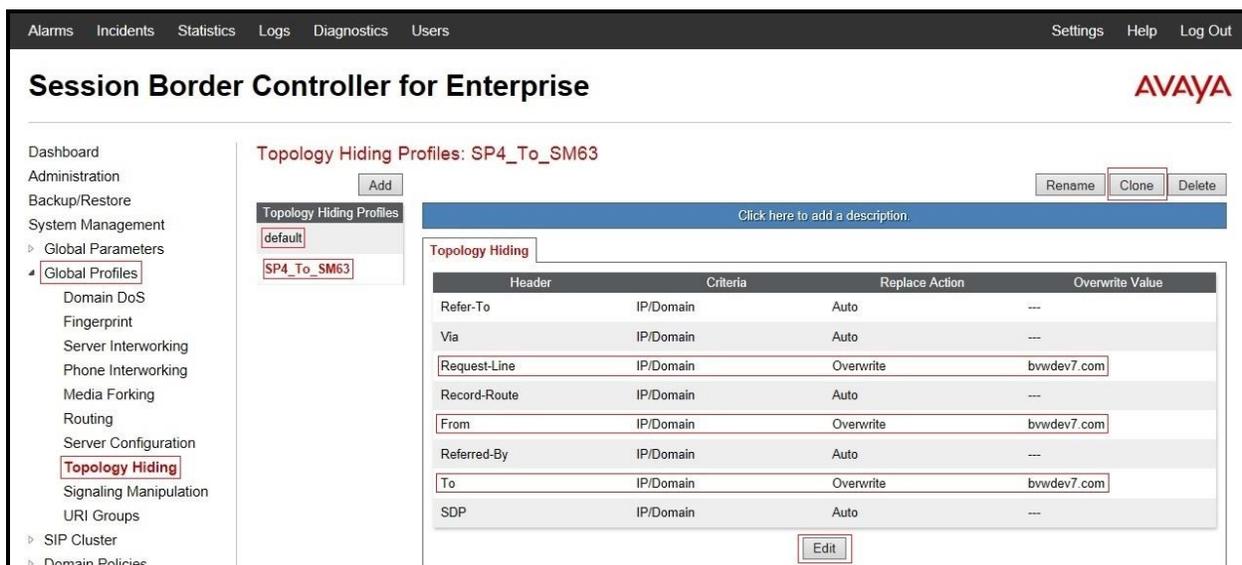
From the menu on the left-hand side, select **Global Profiles** → **Topology Hiding**.

Select **default** under **Topology Hiding Profiles**, and click **Clone**. Enter **Clone Name: SP4\_To\_SM63**. Click **Finish** (not shown).

Select **SP4\_To\_SM63** under **Topology Hiding Profiles**, and click **Edit**.

- For the Header **Request-Line**,
  - In the **Criteria** column select **IP/Domain**.
  - In the **Replace Action** column select: **Overwrite**.
  - In the **Overwrite Value** column: **bwvdev7.com**.
- For the Header **From**,
  - In the **Criteria** column select **IP/Domain**.
  - In the **Replace Action** column select: **Overwrite**.
  - In the **Overwrite Value** column: **bwvdev7.com**.
- For the Header **To**,
  - In the **Criteria** column select **IP/Domain**.
  - In the **Replace Action** column select: **Overwrite**.
  - In the **Overwrite Value** column: **bwvdev7.com**.

Click **Finish** (not shown).



The screenshot shows the Avaya Session Border Controller for Enterprise interface. The main content area is titled "Topology Hiding Profiles: SP4\_To\_SM63". It features a table with the following data:

Header	Criteria	Replace Action	Overwrite Value
Refer-To	IP/Domain	Auto	---
Via	IP/Domain	Auto	---
Request-Line	IP/Domain	Overwrite	bwvdev7.com
Record-Route	IP/Domain	Auto	---
From	IP/Domain	Overwrite	bwvdev7.com
Referred-By	IP/Domain	Auto	---
To	IP/Domain	Overwrite	bwvdev7.com
SDP	IP/Domain	Auto	---

Figure 80 - Topology Hiding Session Manager

## 7.2.9. Configure Topology Hiding – MTS Allstream Site

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

Select **default** under **Topology Hiding Profiles**, and click **Clone**. Enter **Clone Name: SM63\_To\_SP4**. Click **Finish** (not shown).

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes Alarms, Incidents, Statistics, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header displays "Session Border Controller for Enterprise" and the Avaya logo. A left-hand navigation menu lists various configuration areas, with "Global Profiles" and "Topology Hiding" highlighted. The main content area is titled "Topology Hiding Profiles: SM63\_To\_SP4" and features an "Add" button, a list of profiles (default, SP4\_To\_SM63, SM63\_To\_SP4), and buttons for "Rename", "Clone", and "Delete". Below this is a "Topology Hiding" table with columns for Header, Criteria, Replace Action, and Overwrite Value. The table lists several headers such as Refer-To, Via, Request-Line, Record-Route, From, Referred-By, To, and SDP, all with "IP/Domain" as criteria and "Auto" as the replace action. An "Edit" button is located at the bottom of the table.

Header	Criteria	Replace Action	Overwrite Value
Refer-To	IP/Domain	Auto	---
Via	IP/Domain	Auto	---
Request-Line	IP/Domain	Auto	---
Record-Route	IP/Domain	Auto	---
From	IP/Domain	Auto	---
Referred-By	IP/Domain	Auto	---
To	IP/Domain	Auto	---
SDP	IP/Domain	Auto	---

**Figure 81 - Topology Hiding MTS Allstream**

### **7.3. Domain Policies**

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

### 7.3.1. Create End Point Policy Groups

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

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

- Select **Add**.
- Enter **Group Name: SM63\_SP4\_PolicyG**.
  - **Application Rule: default.**
  - **Border Rule: default.**
  - **Media Rule: default-low-med.**
  - **Security Rule: default-med.**
  - **Signaling Rule: default.**
  - **Time of Day: default.**
- Select **Finish** (not shown).

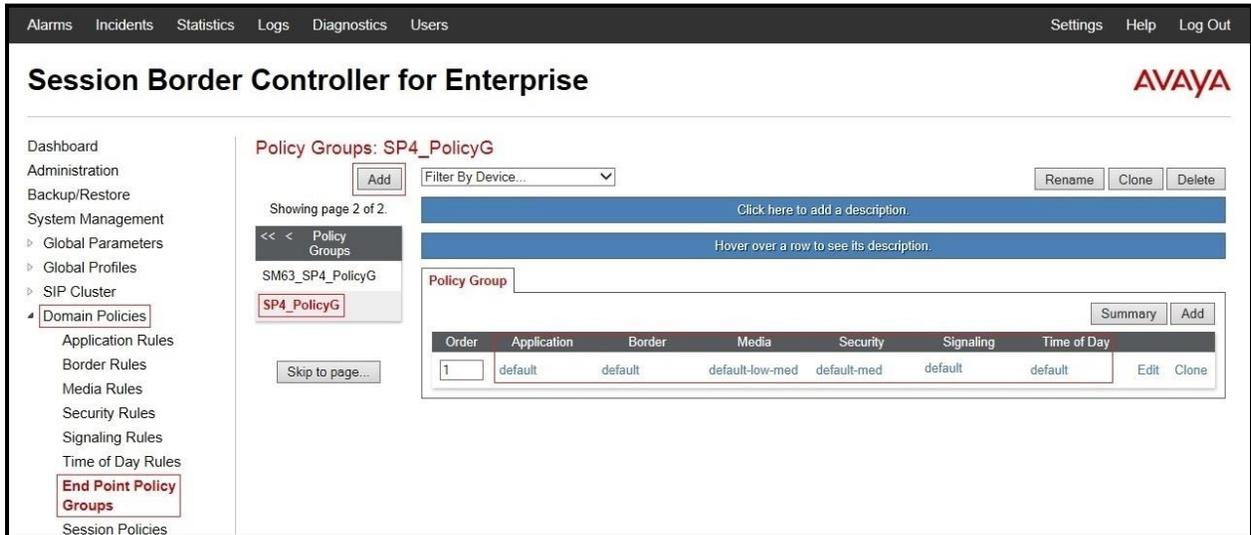
The screenshot displays the Avaya Session Manager web interface. The top navigation bar includes links for Alarms, Incidents, Statistics, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header reads "Session Border Controller for Enterprise" with the AVAYA logo. The left-hand navigation menu is expanded to "Domain Policies" and "End Point Policy Groups". The main content area shows the configuration for a Policy Group named "SM63\_SP4\_PolicyG". It includes an "Add" button, a "Filter By Device..." dropdown, and "Rename", "Clone", and "Delete" buttons. Below this, there is a table of policy rules for the group. The table has columns for Order, Application, Border, Media, Security, Signaling, and Time of Day. The first row shows the following values: Order: 1, Application: default, Border: default, Media: default-low-med, Security: default-med, Signaling: default, Time of Day: default. There are "Edit" and "Clone" buttons for each row. A "Summary" and "Add" button are also present at the bottom right of the table area.

Order	Application	Border	Media	Security	Signaling	Time of Day
1	default	default	default-low-med	default-med	default	default

Figure 82 – Session Manager - End Point Policy Group

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

- Select **Add**.
- Enter **Group Name: SP4\_PolicyG**.
  - **Application Rule: default.**
  - **Border Rule: default.**
  - **Media Rule: default-low-med.**
  - **Security Rule: default-med.**
  - **Signaling Rule: default.**
  - **Time of Day: default.**
- Select **Finish** (not shown).



**Figure 83 - MTS Allstream - End Point Policy Group**

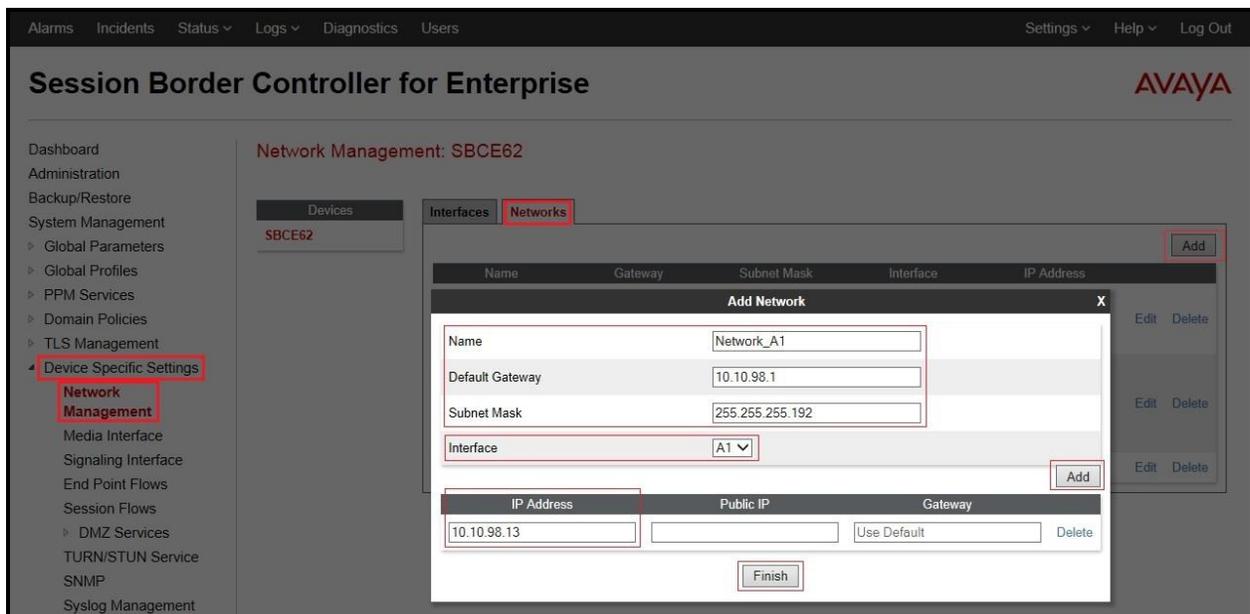
## 7.4. Device Specific Settings

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

### 7.4.1. Manage Network Settings

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

- Select **Networks** tab and click **Add** button to add a network of inside interface as followings:
  - **Name: Network\_A1.**
  - **Default Gateway: 10.10.98.1.**
  - **Subnet Mask: 255.255.255.192.**
  - **Interface: A1** (This is Avaya SBCE inside interface).
  - Click **Add** button to add **IP Address** for inside interface: **10.10.98.13.**
  - Click **Finish** button to save the changes.

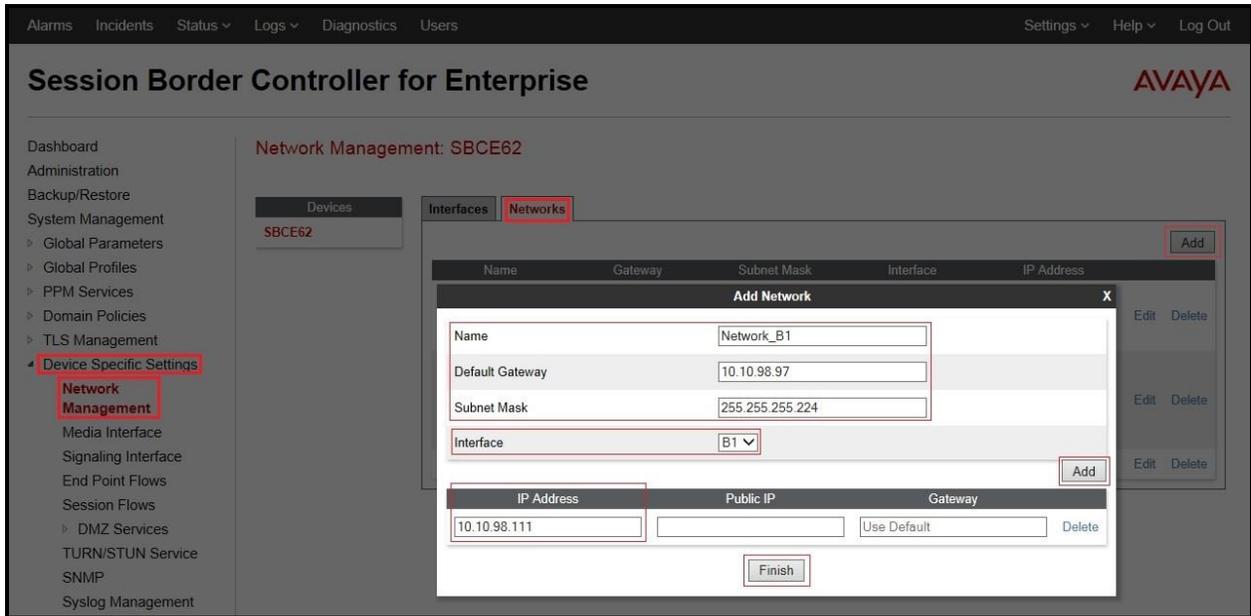


**Figure 84 - Network Management – Inside Interface**

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

- Select **Networks** tab and click **Add** button to add a network of outside interface as followings:
  - **Name: Network\_B1.**
  - **Default Gateway: 10.10.98.97.**
  - **Subnet Mask: 255.255.255.224.**

- **Interface: B1** (This is Avaya SBCE outside interface).
- Click **Add** button to add **IP Address** for outside interface: **10.10.98.111**.
- Click **Finish** button to save the changes.



**Figure 85 - Network Management – Outside Interface**

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

- Select **Interfaces** tab
- Click on the **Status** of the physical interfaces being used and change them to **Enabled** state.

The screenshot shows the Avaya Session Border Controller for Enterprise web interface. The top navigation bar includes Alarms, Incidents, Status, Logs, Diagnostics, Users, Settings, Help, and Log Out. The main header displays "Session Border Controller for Enterprise" and the AVAYA logo. A left-hand navigation menu lists various system management options, with "Device Specific Settings" and "Network Management" highlighted. The main content area is titled "Network Management: SBCE62" and features two tabs: "Devices" and "Interfaces". The "Interfaces" tab is active, displaying a table with the following data:

Interface Name	VLAN Tag	Status
A1		Enabled
A2		Disabled
B1		Enabled
B2		Disabled

An "Add VLAN" button is located in the top right corner of the interface table.

**Figure 86 - Network Interfaces Status**

## 7.4.2. Create Media Interfaces

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

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

- Select **Add**.
  - **Name: InsideMedia1.**
  - **Media IP: 10.10.98.13** (Avaya SBCE Internal IP Address toward Session Manager).
  - **Port Range: 35000 – 40000.**
  - Click **Finish** (not shown).
- Select **Add**.
  - **Name: OutsideMedia1.**
  - **Media IP: 10.10.98.111** (Avaya SBCE External IP Address toward MTS Allstream SIP Trunk Service).
  - **Port Range: 35000 – 40000.**
  - Click **Finish** (not shown).

Session Border Controller for Enterprise

Media Interface: SBCE62

Modifying or deleting an existing media interface will require an application restart before taking effect. Application restarts can be issued from System Management.

Name	Media IP	Port Range	Edit	Delete
InsideMedia1	10.10.98.13	35000 - 40000	Edit	Delete
OutsideMedia1	10.10.98.111	35000 - 40000	Edit	Delete

Add

**Figure 87 - Media Interface**

### 7.4.3. Create Signaling Interfaces

Signaling Interfaces define the type of signaling on the ports.

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

- Select **Add**.
  - **Name: InsideUDP1.**
  - **Media IP: 10.10.98.13** (Avaya SBCE Internal IP Address toward Session Manager).
  - **UDP Port: 5060.**
  - Click **Finish** (not shown).
- Select **Add**.
  - **Name: OutsideUDP1.**
  - **Media IP: 10.10.98.111** (Avaya SBCE External IP Address toward MTS Allstream SIP Trunk Service).
  - **UDP Port: 5060.**
  - Click **Finish** (not shown).

Alarms Incidents Status Logs Diagnostics Users Settings Help Log Out

## Session Border Controller for Enterprise

AVAYA

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

Devices  
SBCE62

Signaling Interface

Modifying or deleting an existing signaling interface will require an application restart before taking effect. Application restarts can be issued from System Management.

Name	Signaling IP	TCP Port	UDP Port	TLS Port	TLS Profile		
InsideUDP1	10.10.98.13	---	5060	---	None	Edit	Delete
OutsideUDP1	10.10.98.111	---	5060	---	None	Edit	Delete

Add

**Figure 88 - Signaling Interface**

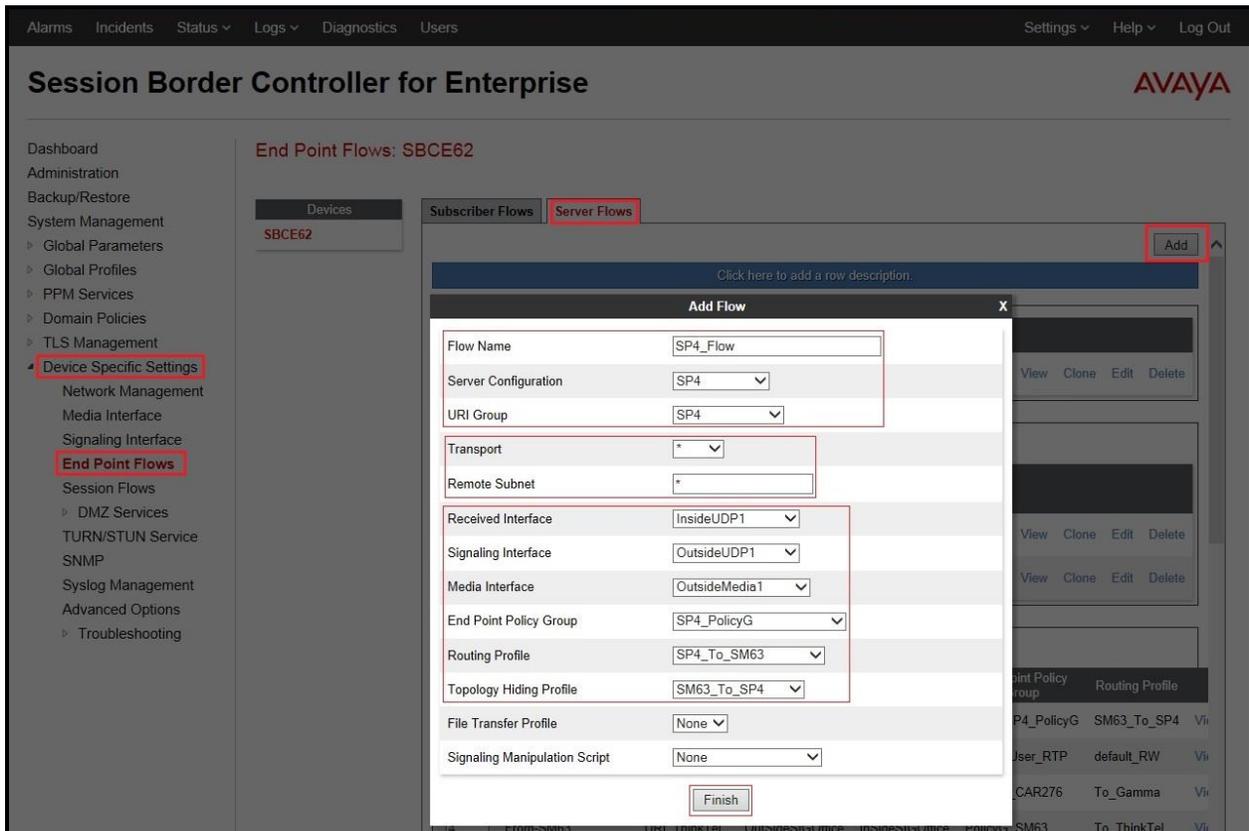
#### **7.4.4. Configuration End Point Flows**

Endpoint flows are used to determine the signaling endpoints involved in a call in order to apply the appropriate policies. When a packet arrives at the Avaya SBCE, the content of the packet (IP addresses, URIs, etc) is used to determine which flow it matches. Once the flow is determined, the flow points to policies and profiles which control processing, privileges, authentication, routing, etc. Once routing is applied and the destination endpoint is determined, the policies for the destination endpoint are applied. Thus, two flows are involved in every call: the source endpoint flow and the destination endpoint flow. In the case of the compliance test, the signaling endpoints are Session Manager and the MTS Allstream SIP Trunk Service.

### 7.4.4.1 Create End Point Flows – Session Manager Flows

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

- Select the **Server Flows** tab.
- Select **Add**, enter **Flow Name: SP4\_Flow**.
  - **Server Configuration: SP4** (refer to **Section 7.2.5**).
  - **URI Group: SP4** (refer to **Section 7.2.3**).
  - **Transport: \***.
  - **Remote Subnet: \***.
  - **Received Interface: InsideUDP1** (refer to **Section 7.4.3**).
  - **Signaling Interface: OutsideUDP1** (refer to **Section 7.4.3**).
  - **Media Interface: OutsideMedia1** (refer to **Section 7.4.2**).
  - **End Point Policy Group: SP4\_PolicyG** (refer to **Section 7.3.1**).
  - **Routing Profile: SP4\_To\_SM63** (refer to **Section 7.2.6**).
  - **Topology Hiding Profile: SM63\_To\_SP4** (refer to **Section 7.2.9**).
  - Click **Finish**.



**Figure 89 - End Point Flows 1**

### 7.4.4.2 Create End Point Flows – Trunk Flows

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

- Select the **Server Flows** tab.
- Select **Add**, enter **Flow Name: SM63\_Flow**.
  - **Server Configuration: SM63** (Refer to Section 7.2.4).
  - **URI Group: SP4** (Refer to Section 7.2.3).
  - **Transport: \***.
  - **Remote Subnet: \***.
  - **Received Interface: OutsideUDP1** (refer to Section 7.4.3).
  - **Signaling Interface: InsideUDP1** (refer to Section 7.4.3).
  - **Media Interface: InsideMedia1** (refer to Section 7.4.2).
  - **End Point Policy Group: SM63\_SP4\_PolicyG** (refer to Section 7.3.1).
  - **Routing Profile: SM63\_To\_SP4** (refer to Section 7.2.7).
  - **Topology Hiding Profile: SP4\_To\_SM63** (refer to Section 7.2.8).
  - Click **Finish**.

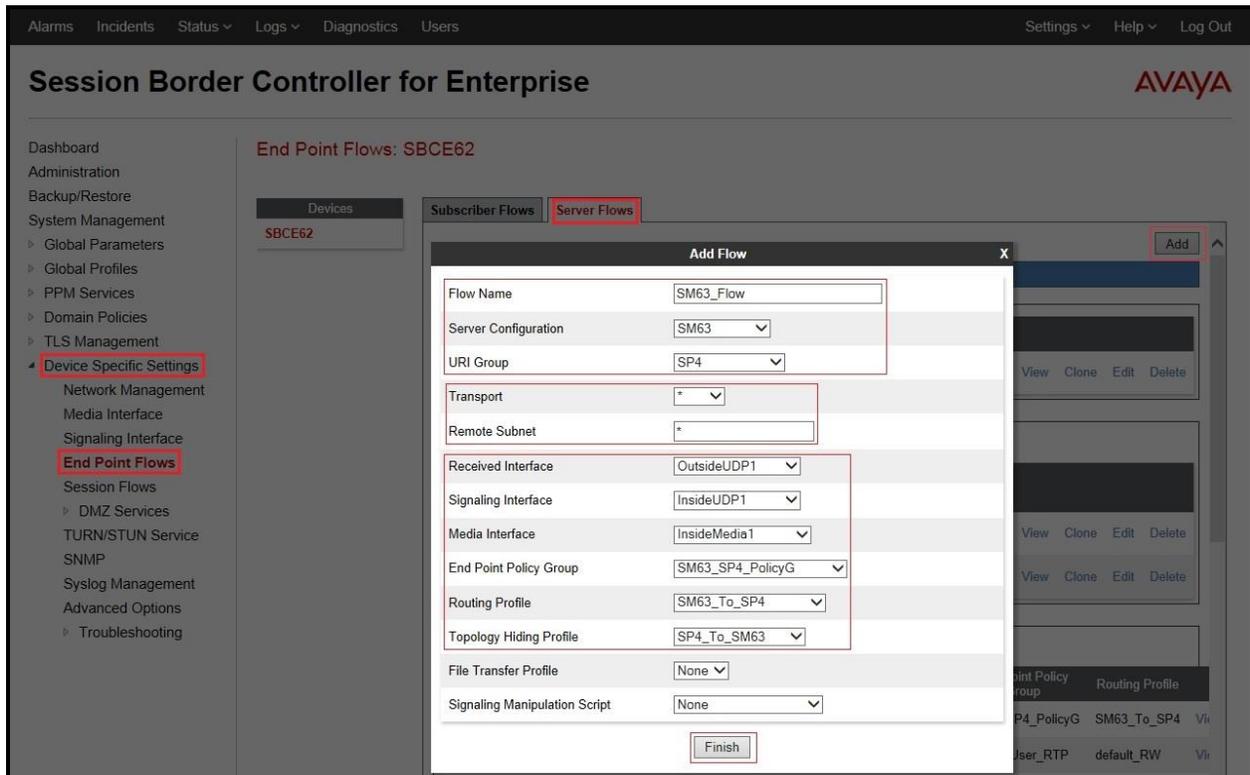


Figure 90 - End Point Flows 2

## **8. MTS Allstream SIP Trunk Service Configuration**

MTS Allstream is responsible for the network configuration of the MTS Allstream SIP Trunk Service. MTS Allstream SIP Trunk Service will require that the customer provide the public IP address used to reach the Avaya SBCE public interface at the edge of the enterprise. MTS Allstream SIP Trunk Service will provide the IP addresses of MTS Allstream's SIP proxy/SBC, IP addresses of media sources and Direct Inward Dialed (DID) numbers assigned to the enterprise. This information is used to complete configurations for CS1000, Session Manager, and Avaya SBCE discussed in the previous sections.

The configuration between MTS Allstream SIP Trunk Service and the enterprise is a static configuration. There is no authentication of the SIP trunk from enterprise users to the MTS Allstream's network.

## 9. Verification Steps

The following steps may be used to verify the configuration.

### 9.1. 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.

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

#### Active Call Trace (Id 80)

The following is an example of one of the commands available on the CS1000 to trace the DN for which the call is in progress or idle (1257). The call scenario involved PSTN phone number 1613XXX5206 calling 647XXX1257 (which is translated to phone 1257).

- Login into CS1000 Signaling Server 10.10.97.177 with admin account and password.
- Issue a command “cslogin” to login on to the CS1000 Call Server.
- Log in to the Overlay command prompt, issue the command **ld 80** and then **trace 0 1257**.
- After the call is released, issue command **trac 0 1257** again to see if the DN is released back to idle state.

Below is the actual output of the CS1000 Call Server Command Line mode when the **1257** is in call state:

```
>ld 80
TRA000
.trac 0 1257

ACTIVE VTN 096 0 00 02

ORIG VTN 100 0 01 00 VTRK IPTI RMBR 101 1 INCOMING VOIP GW CALL
FAR-END SIP SIGNALLING IP: 10.10.98.13
FAR-END MEDIA ENDPOINT IP: 10.10.98.13 PORT: 36194
FAR-END VendorID: AVAYA-SM-6.3.7.0.637008
TERM VTN 096 0 00 02 KEY 0 SCR MARP CUST 0 DN 1257 TYPE 2002P2
SIGNALLING ENCRYPTION: INSEC
MEDIA ENDPOINT IP: 10.33.5.23 PORT: 5200
MEDIA PROFILE: CODEC G.711 MU-LAW PAYLOAD 20 ms VAD OFF
RFC2833: RXPT 101 TXPT 101 DIAL DN 1257
MAIN_PM ESTD
TALKSLOT ORIG 6 TERM 11
EES_DATA:
NONE
QUEU NONE
CALL ID 501 80

---- ISDN ISL CALL (ORIG) ----
CALL REF # = 385
BEARER CAP = VOICE
HLC =
```

```
CALL STATE = 10  ACTIVE
CALLING NO = 1613XXX5206 NUM_PLAN:UNKNOWN  TON:UNKNOWN  ESN:UNKNOWN
CALLED NO = 647XXX1257 NUM_PLAN:UNKNOWN  TON:UNKNOWN  ESN:UNKNOWN
```

And this is the example after the call to 1257 is finished.

```
>ld 80
TRA000
.trac 0 1257
IDLE VTN 96 0 00 02  MARP
```

### **SIP Trunk monitoring (ld 32)**

Place a call inbound from PSTN (1613XXX5206) to an internal device (647XXX1257). Then check the SIP trunk status by using ld 32, one trunk is BUSY.

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

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

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

### 9.3. Protocol Trace

Below is a wireshark trace of the same call scenario described in **Section 9.2**. This capture was taken by Wireshark application on a computer connected to the same subnet with the public interface of Avaya SBCE.

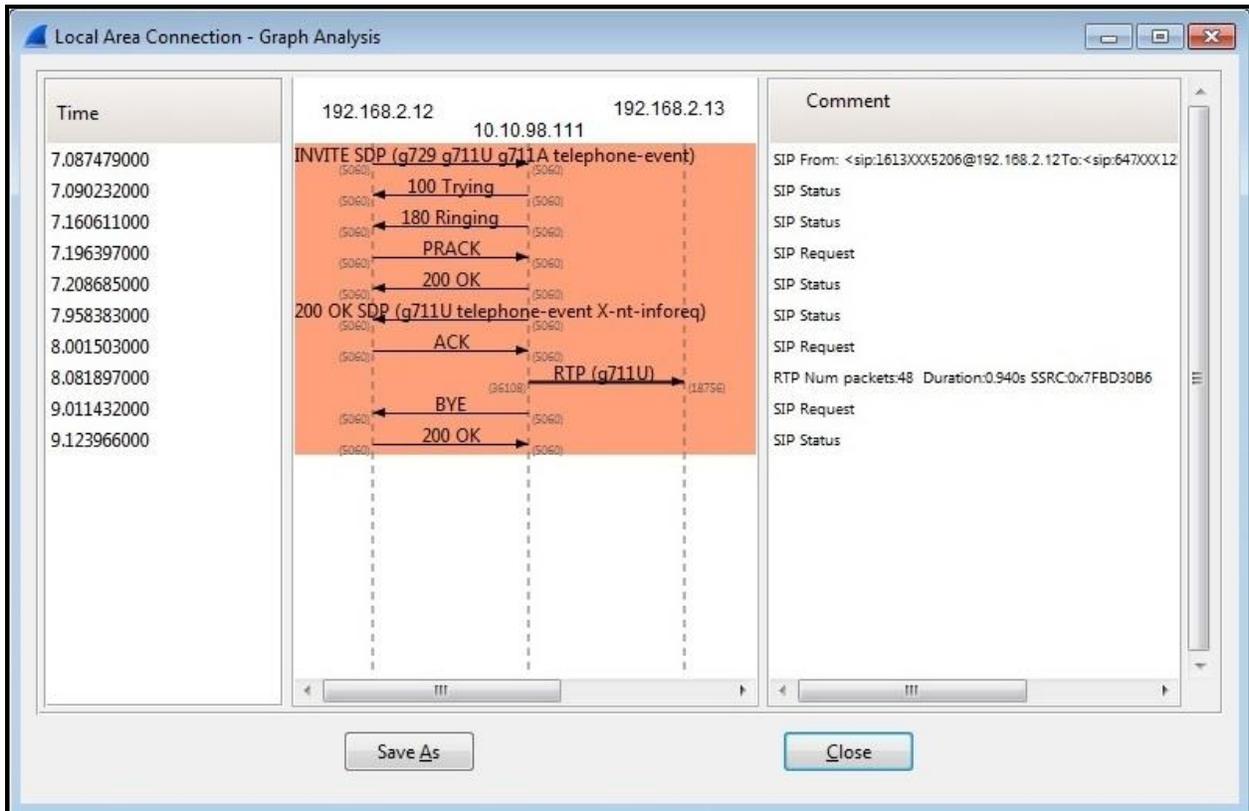


Figure 91 – SIP Call Trace

## 10. Conclusion

All of the test cases have been executed. Despite observations seen during the testing, as noted in **Section 2.2**, the test met the objectives outlined in **Section 2.1**. The MTS Allstream SIP Trunk Service is considered **compliant** with Avaya Communication Server 1000 Release 7.6, Avaya Aura<sup>®</sup> Session Manager Release 6.3 and Avaya Session Border Controller for Enterprise Release 6.3.

## 11. References

This section references the documentation relevant to these Application Notes.

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

### Avaya Communication Server 1000

- [1] *Network Routing Service Fundamentals, Avaya Communication Server 1000*, Release 7.6, Document Number NN43001-130, Issue 04.01, March 2013
- [2] *IP Peer Networking Installation and Commissioning, Avaya Communication Server 1000*, Release 7.6, Document Number NN43001-313, Issue 06.01, March 2013
- [3] *Communication Server 1000E Overview, Avaya Communication Server 1000*, Release 7.6, Document Number NN43041-110, Issue 06.01, March 2013
- [4] *Unified Communications Management Common Services Fundamentals, Avaya Communication Server 1000*, Release 7.6, Document Number NN43001-116, Issue 06.01, March 2013
- [5] *Dialing Plans Reference, Avaya Communication Server 1000*, Release 7.6, Document Number NN43001-283, Issue 06.01, March 2013.
- [6] *Product Compatibility Reference, Avaya Communication Server 1000*, Release 7.6, Document Number NN43001-256, Issue 06.01 Standard, March 2013

### Avaya Aura<sup>®</sup> Session Manager/System Manager

- [7] *Administering Avaya Aura<sup>®</sup> Session Manager*, Release 6.3, Issue 7, September 2014
- [8] *Maintaining and Troubleshooting Avaya Aura<sup>®</sup> Session Manager*, Release 6.3, Issue 6, December 2014
- [9] *Administering Avaya Aura<sup>®</sup> System Manager*, Release 6.3, Issue 2, May 2013

### Avaya Session Border Controller for Enterprise

- [10] *Avaya Session Border Controller for Enterprise Overview and Specification*, Release 6.3, Issue 3, October 2014
- [11] *Administering Avaya Session Border Controller for Enterprise*, Release 6.2, Issue 2, January 2014

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