Avaya Solution & Interoperability Test Lab

# Application Notes for Configuring Level 3 SIP Trunking Service with Avaya Communication Server 1000 Release 7.5, Avaya Aura® Session Manager Release 6.1 and Acme Packet Session Border Controller Release 6.2 – Issue 1.0

## Abstract


These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between Level 3 SIP Trunking Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of an Avaya Communication Server 1000 7.5, Avaya Aura® Session Manager 6.1, Acme Packet Session Border Controller 6.2 and various Avaya endpoints. This documented solution does not extend to configurations without Avaya Aura® Session Manager and Acme Packet Session Border Controller.

Level 3 is a member of the Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing is conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

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# 1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between Level 3 SIP Trunking Service (Level 3) and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of an Avaya Communication Server 1000 (CS1000) 7.5, Avaya Aura® Session Manager 6.1, Acme Packet Session Border Controller (Acme SBC) 6.2 and various Avaya endpoints.

Customers using this Avaya SIP-enabled enterprise solution with Level 3 are able to place and receive PSTN calls via a broadband connection. This converged network solution is an alternative to traditional PSTN trunking such as analog and/or ISDN-PRI.

# 2. General Test Approach and Test Results

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.

Level 3 is a member of the Avaya DevConnect Service Provider program. The general test approach is to connect a simulated enterprise to Level 3 via the public internet and exercise the features and functionality listed in **Section 2.1**.

## 2.1. Interoperability Compliance Testing

To verify Level 3 SIP Trunking Service interoperability, the following features and functionalities are covered during the compliance test:

- Incoming PSTN call to various phone types including SIP, UNIStim, PC2050 softphone, digital and analog telephone at the enterprise. All inbound calls from PSTN are routed to the enterprise across the SIP trunk from the service provider.
- Outgoing PSTN call from various phone types including SIP, UNIStim, PC2050 softphone, digital and analog telephone at the enterprise. All outbound calls to PSTN are routed from the enterprise across the SIP trunk to the service provider.
- Dialing plans including local, long distance, international, outbound toll-free, operator assisted calls, local directory assistance (411)… etc.
- Proper Codec Negotiation with G.729 and G.711MU codecs.
- Proper Early Media transmission with G.729 and G.711MU codecs.
- Incoming and outgoing fax over IP with T.38 codec.
- DTMF tone transmissions as out-of-band RTP event as per RFC2833.
- Caller ID presentation and Caller ID restriction.
- Response to incomplete call attempts and trunk errors.
- Voicemail navigation for inbound and outbound calls.
- User features such as hold and resume, transfer and conference.

- Off-net call transfer with SIP re-INVITE method.
- Off-net call forward with SIP Diversion method.
- SIP Digest Authentication.

Items are not supported or not tested including the following:

- Inbound toll-free and outbound emergency calls (911) are supported but are not tested as part of the compliance test because Level 3 does not provide the necessary configuration.
- Session Timer refresh is not supported.
- Reliable of Provisional Responses (RFC3262) is not supported.
- Off-net call forwarding using History-Info method is not supported.

## 2.2. Test Results

Interoperability testing of Level 3 SIP Trunking Service with the Avaya SIP-enabled enterprise solution is completed with successful results for all test cases with the exception of the observations/limitations described below.

1. **Fax over IP using G.711 codec is not recommended**. Transmitting fax over IP using a G.711 codec appears to work for regular fax machines. However, when using an integrated fax modem on a PC, the fax call fails as Level 3 unexpectedly attempts to switch the codec from G.711MU to T.38. This is a known issue on Level 3 SIP Trunking Service but there is no resolution available at this time.

2. **In blind transfer off-net scenario, the calling PSTN does not hear ringback tone when the called PSTN is ringing**. This limitation is encountered when performing a work around to support a blind transfer call without an UPDATE/SDP method. Before completing the transferred call, the CS1000 uses an UPDATE/SDP method to anchor ring back tone on the 2$^{nd}$ leg to the 1$^{st}$ leg. However, Level 3 does not appear to support this method, it rejects the UPDATE/SDP with a "500 Internal Server Error" response. A workaround has been made to eliminate the UPDATE method on inbound signaling, that makes the CS1000 automatically disable UPDATE from being sent to Level 3. This approach is achieved by additional configuration made to the Acme SBC and the CS1000 as described below:
   - On the Acme SBC, create a Header Manipulation Rule (HMR) to delete UPDATE in the Allow header on inbound signaling. For a detailed configuration, please refer to **Section 7.7**, sip-manipulation rule **Level3_To_CS1K**, header-rule **manipAllow**.
   - On the CS1000, enable plug-in 501 in pdt mode. The CS1000 deactivates the blind transfer feature when the far end does not support UPDATE. To reactivate blind transfer functionality, the plug-in 501 has to be enabled. For a detailed configuration, please refer to **Section 5.5.10**.

   **Note**: The CS1000 requires support of UPDATE, but Level3 does not support this method. Not supporting UPDATE may result in significant service degradation and feature breakage.

3. **Off-net call transfer, the calling party name and number is not updated to calling PSTN party** When the CS1000 transfers an incoming call off-net to the PSTN, it sends a 200OK with the true connected calling party name and number in Remote-Party-ID

header to the calling PTSN. However, the calling party name and number are not updated; the calling PTSN party still displays the calling party number of the CS1000. This is a known issue on Level 3 SIP Trunking Service. It is recommended that Level 3 should support the calling party information update. The feature also needs to be supported by the service provider hosting the calling PSTN party. This issue has low user impact, it is listed here simply as an observation.

4. **CS1000 SIP phone transfer off-net to the PSTN is not successful with Music On Hold enabled**. In an inbound or outbound call between a CS1000 SIP phone and PSTN_1, the CS1000 SIP phone performs an off-net transfer back to PSTN_2. The transfer fails. PSTN_1 continues to hear ringing after the call has been answered by PSTN_2. This call scenario is successful with other endpoints .e.g. UNIStim or digital phones. The issue does not happen when Music On Hold is disabled. An internal tracking number wi01017194 has been created. This issue is simply listed here as a limitation.

5. **CS1000 phone holds and retrieves an outbound call causing the calling party number to be changed**. After retrieving a call, the calling party number previously displayed on CS1000 phone will be replaced by Route ACOD – Trunk Channel ID. This is a known issue on the CS1000 but there is no resolution available at this time. This issue has low user impact, it is listed here simply as an observation.

6. **CS1000 SIP phone calls a local UNIStim phone then blind transfers to PSTN causing the calling party number to be changed.** The call is successfully transferred. However, the UNIStim phone displays Route ACOD – Trunk Channel ID instead of displaying the PSTN calling party name and number. This is a known issue on the CS1000 but there is no resolution available at this time. This issue has low user impact, it is listed here simply as an observation.

7. **Digest Authentication on inbound call is corrected and works properly**. When the CS1000 holds an inbound call, the re-INVITE from the CS1000 is challenged by a 401 from Level 3 to do Digest Authentication. However, the CS1000 does not resend another re-INVITE with Authorization header as expected. The issue has been corrected by applying patch cs1000-vtrk-7.50.17.16-30.i386.000 to the CS1000 SIP Trunk Gateway.

## 2.3. Support

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

For technical support on Level 3 SIP Trunking Service, please contact Level 3 technical support at:

- Phone: 1-877-453-8353)
- Website: http://www.level3.com/en/contact-us/

# 3. Reference Configuration

**Figure 1** illustrates the sample Avaya SIP-enabled enterprise solution connected to the Level 3 SIP Trunking Service (Vendor Validation circuit) through a public Internet WAN connection.

For security purposes, the real public IP addresses and PSTN routable phone numbers used in the compliance test are not shown in these Application Notes.

Located at the edge of the enterprise network is an Acme SBC. It has a public side that connects to Level 3 via the internet and a private side that connects to the enterprise network. All SIP and RTP traffic entering or leaving the enterprise network flows through the Acme SBC which can protect the enterprise against any outside SIP-based attacks. The Acme SBC provides network address translation at both the IP and SIP layers. The transport protocol between the Acme SBC and Level 3 across the public network is UDP; the transport protocol between the Acme SBC and Session Manager across the enterprise network is TCP. In the compliance testing, the Avaya CPE environment is configured with SIP domain **level3.com** for the enterprise. The Acme SBC is used to adapt the enterprise SIP domain to the IP address based URI-Host known to Level 3. **Figure 1** below illustrates the network diagram for the enterprise.

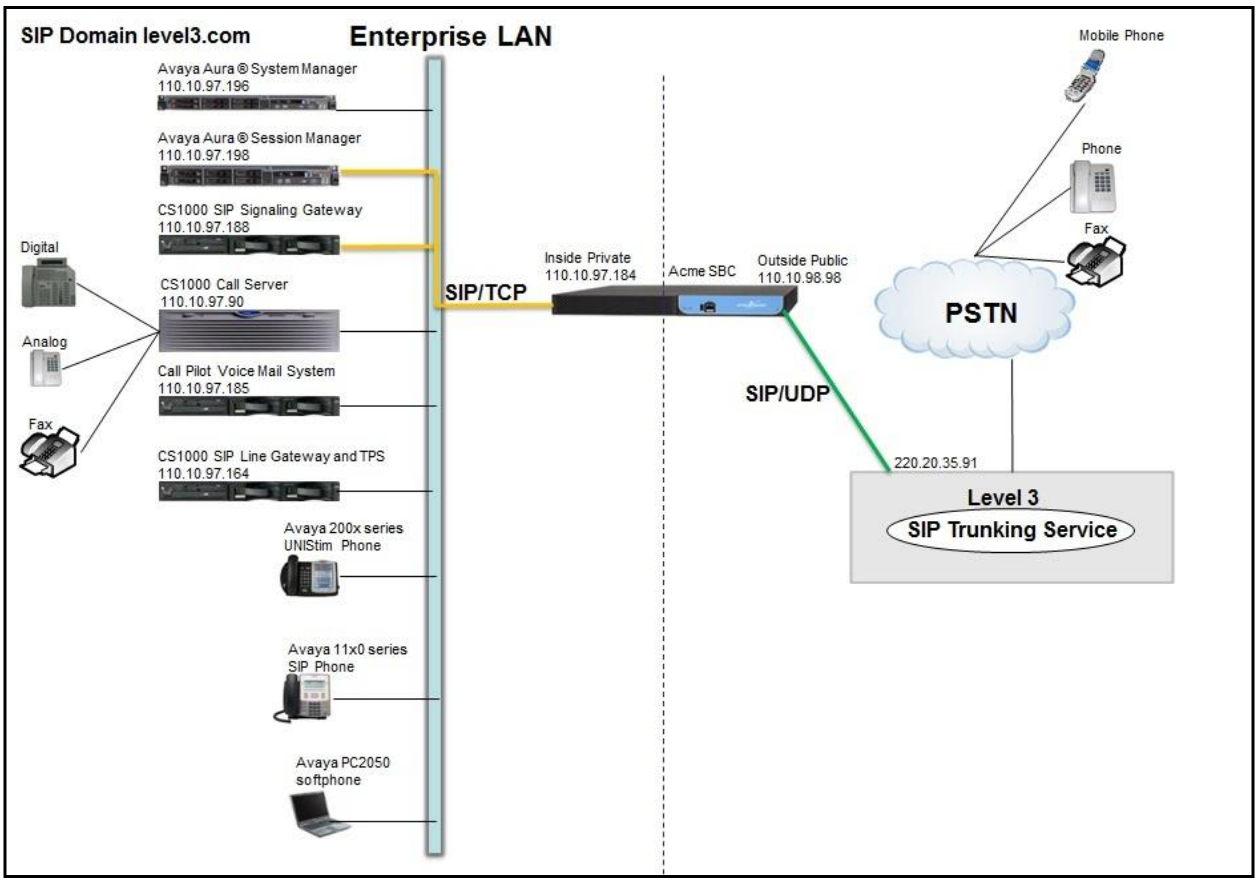



**Figure 1: Avaya IP Telephony Network connecting to Level 3 SIP Trunking Service**

# 4. Equipment and Software Validated

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

| **Avaya IP Telephony Solution Components** | |
|---|---|
| Component | Release |
| Avaya CS1000 7.5 (CPPM) | • Call Server: 7.50 Q GA plus latest DEPLIST – Issue: 01 Release: x2107.50, 2012-05-16 12:51:18 (est)<br>• SSG Server: 7.50.17 GA plus latest Service_Pack_Linux_7.50_17_20120516.ntl<br>• SLG Server: 7.50.17 GA plus latest Service_Pack_Linux_7.50_17_20120516.ntl |
| Avaya Aura® System Manager running on Avaya S8800 Server | • 6.1.5.0<br>Build number 6.1.0.0.7345 Patch 6.1.5.9 |
| Avaya Aura® Session Manager running on Avaya S8800 Server | • 6.1.1.0.611023 |
| Avaya IP Telephone | • 2002 p2: 0604DCJ (UNIStim)<br>• 2004 p2: 0604DCJ (UNIStim)<br>• 1140: 0625C6O (UNIStim)<br>• 1120: 0624C6O (UNIStim)<br>• 2007: 0621C6M (UNIStim)<br>• 1220: 062AC6O (UNIStim)<br>• SIP 1120, 1140: SIP12x0e04.00.04.00<br>• SIP 1220,1240: SIP12x0e04.00.04.00 |
| Avaya CallPilot | 05.00.41.141 |
| Avaya 2050PC softphone | 3.4 |
| Avaya Digital Telephone | n/a |
| Avaya Analog Telephone | n/a |
| Acme Packet Session Border Controller 3800 | Net-Net 3800 Firmware SCX6.2.0 MR-9 GA |
| **Level 3 SIP Trunking Service Components** | |
| Component | Release |
| Level 3 Enterprise Edge | Version 1 |

**Table 1: Equipment and Software Tested**

# 5. Avaya Communication Server 1000 Configuration

This section describes the procedure for configuring the CS1000 for inter-operating with the Level 3.

A two-way SIP trunk is created between the CS1000 and Session Manager to carry traffic to and from the service provider. For an inbound call, the call flows from Level 3 to the Acme SBC to the CS1000 via Session Manager. Once the call arrives at the CS1000, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. Outbound calls to the PSTN are first processed by the CS1000 for outbound feature treatment such as route selection and class of service. Once the CS1000 has selected the proper SIP trunk, the call is routed to Session Manager and then on toward the Acme SBC for egress to the Level 3.

For the compliance test, CS1000 sends 11 digits in the destination headers (e.g., Request-URI and To) and sends 10 digit in the source headers (e.g., From, Contact, and P-Asserted-Identity (PAI)). Level 3 sends 10 digits in destination headers and sends 10 digits in source headers.

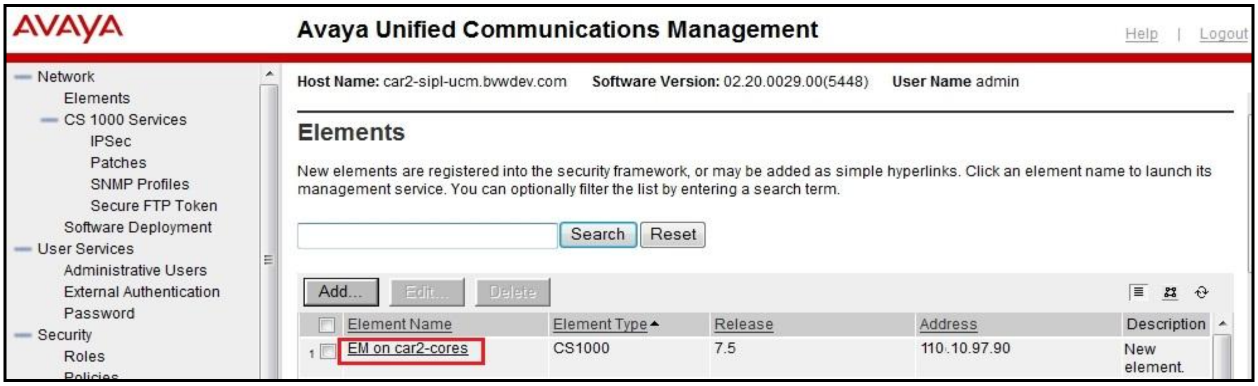
These Application Notes assume the basic configuration has already been administered and is not discussed here. For further information on CS1000, please consult references in **Section 11**.

## 5.1. Login to CS1000

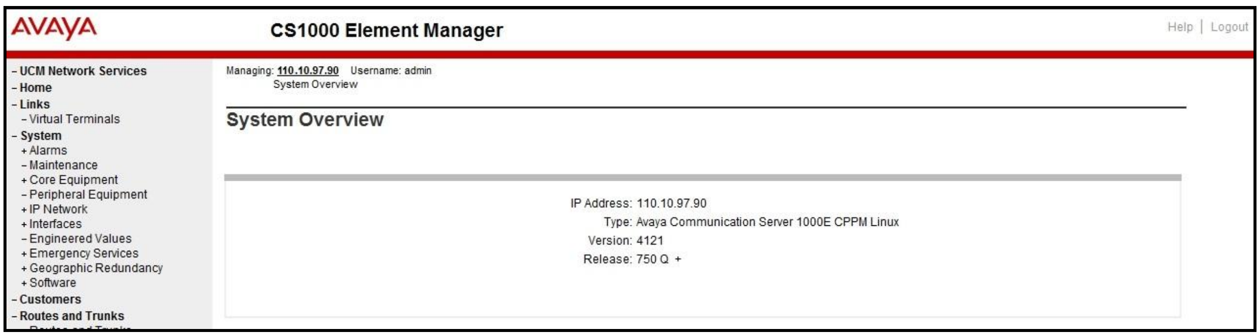
### 5.1.1. Login Unified Communications Management (UCM) and Element Manager (EM)

a) Open web browser and connect to the UCM GUI https://<UCM IP address> as shown in the screenshot below then log in using an appropriate username and password.


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




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




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

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

```
login as: admin

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

admin@110.10.97.172's password:
Last login: Wed Nov  2 11:32:26 2011 from 110.10.98.105

[admin@car2-sps ~]$
[admin@car2-sps ~]$ cslogin

SEC054 A device has connected to, or disconnected from, a pseudo tty without
authentica
ting

TTY 09 SCH MTC TRF BUG OSN   14:05
OVL111 BKGD  44
```

```
OVL111   TTY 10       0     ADMIN
>
```

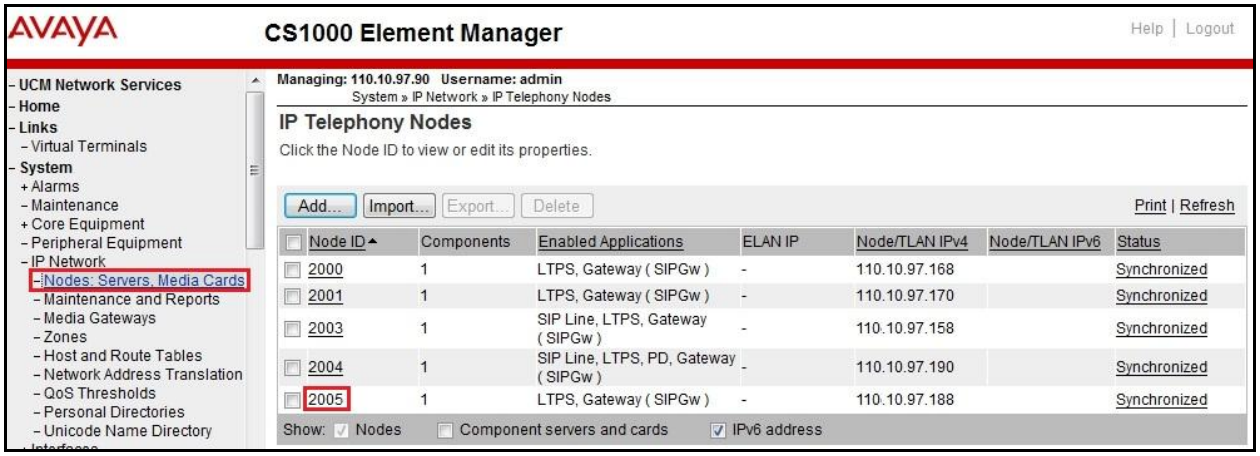
## 5.2. Administer a Node IP Telephony

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

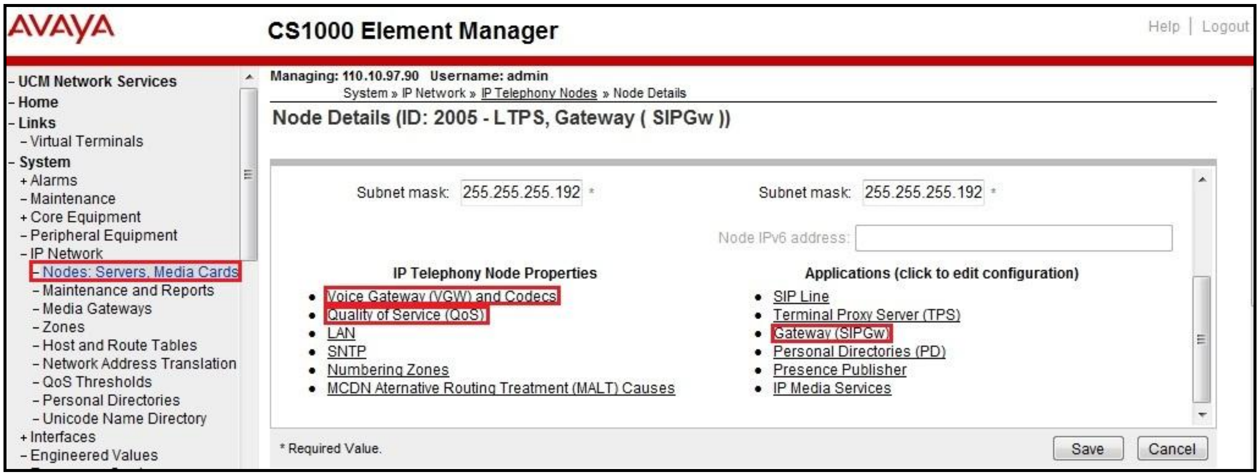
### 5.2.1. Obtain Node IP address

These Application Notes assume the basic configuration has already been administered and that a Node has already been created. This section describes configuration steps for Node ID 2005.

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

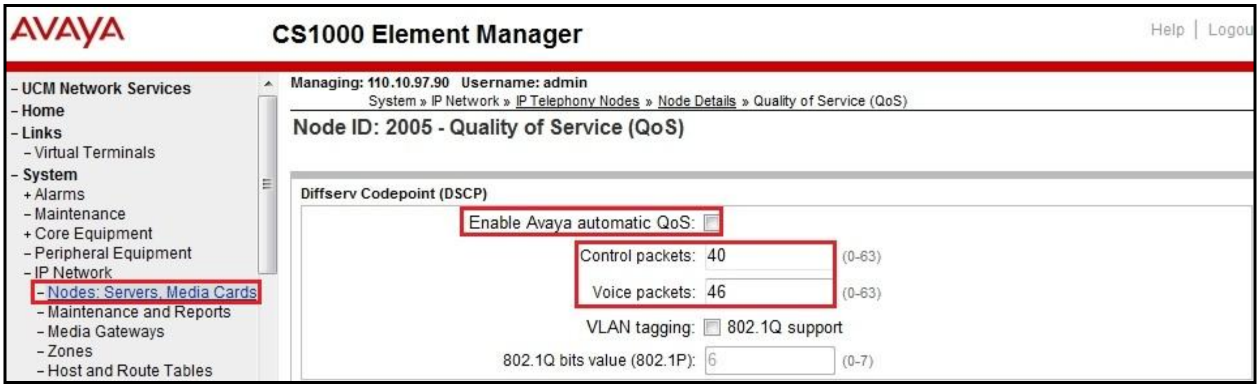



b) The **Node Details** page is shown in the screenshot below with the IP address of the Node ID 2005. The SIP Signaling Gateway uses the **Node IP Address** to connect to Session Manager for the SIP Trunk to Level 3.



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

c) Continued from **Section 5.2.1**.  On the **Node Details** page, select the **Quality of Service (QoS)** link. The default Diffserv values are shown in the screenshot below. Then click the **Save** button.



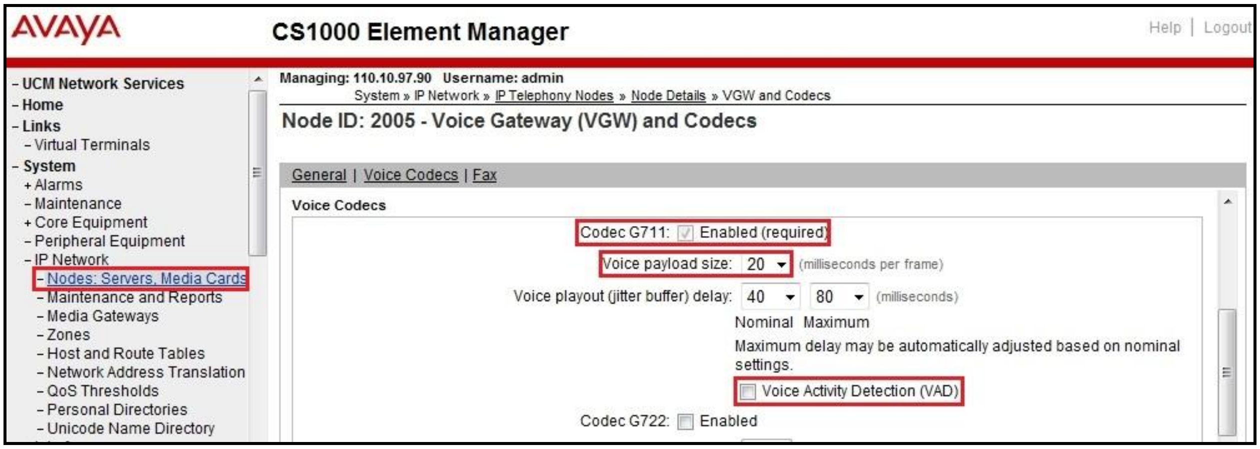

### 5.2.3. Synchronize the new configuration

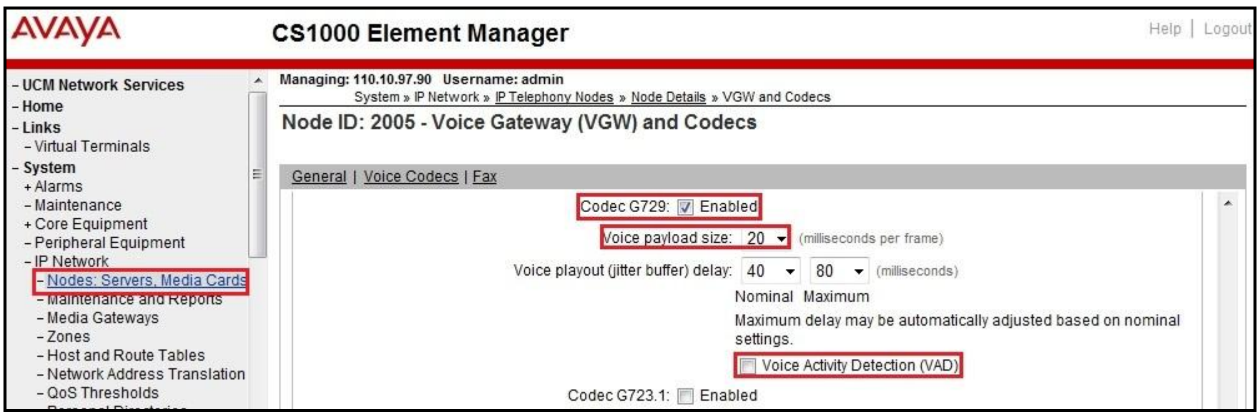
d) Continued from **Section 5.2.3**, return to the **Node Details** page (not shown) and click the **Save** button.
e) The **Node Saved** screen is displayed. Click **Transfer Now** button (not shown).
f) The **Synchronize Configuration Files** screen is displayed (not shown). Check the Signaling Server checkbox and click the **Start Sync** button (not shown).
g) When the synchronization completes, check the Signaling Server check box and click the **Restart Applications** button (not shown).

## 5.3. Administer Voice Codec

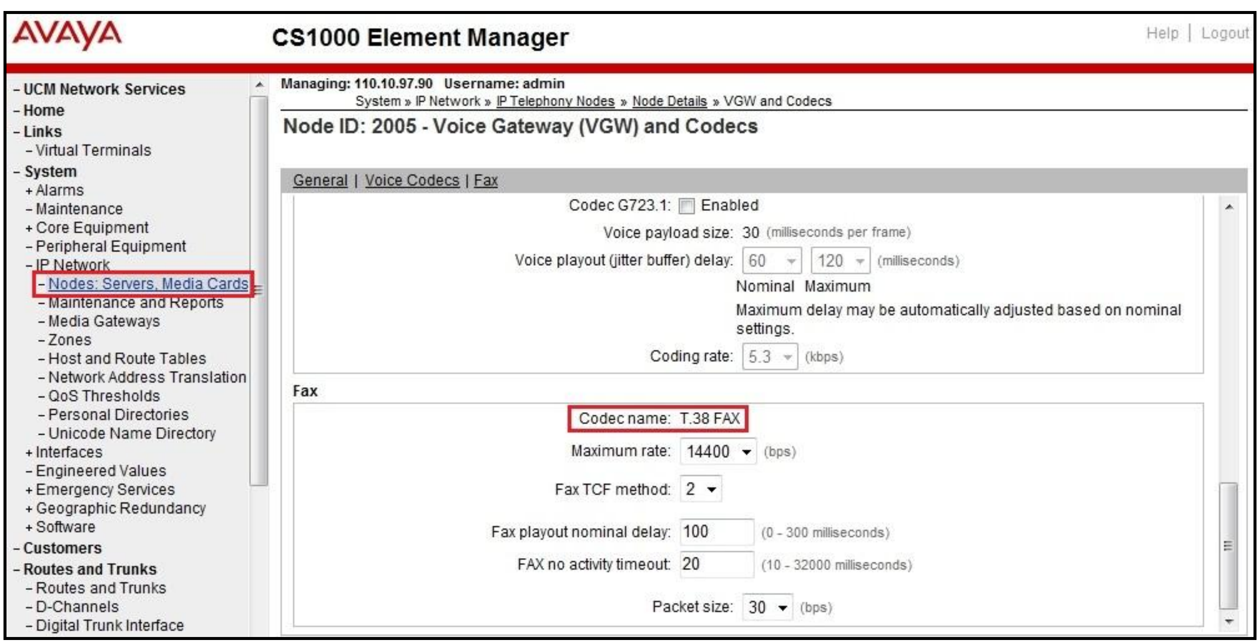
### 5.3.1. Enable Voice Codec, Node IP Telephony

a) To configure a Voice Codec, select **IP Network → Nodes: Servers, Media Cards** from the left pane, and in the **IP Telephony Nodes** screen, select the **Node ID** of the CS1000 system. The **Node Details** screen is displayed as described in **Section 5.2.1**.
b) On the **Node Details** page (not shown), click on **Voice Gateway (VGW) and Codec.**
c) Level 3 supports voice codecs G.729 and G.711, payload size 20 ms, with VAD disabled. The following screenshots show appropriated voice codec profiles configured on the CS1000.




d) For Fax over IP, Level 3 supports T.38. This parameter is enabled by default on the CS1000 as shown in the following screenshot.

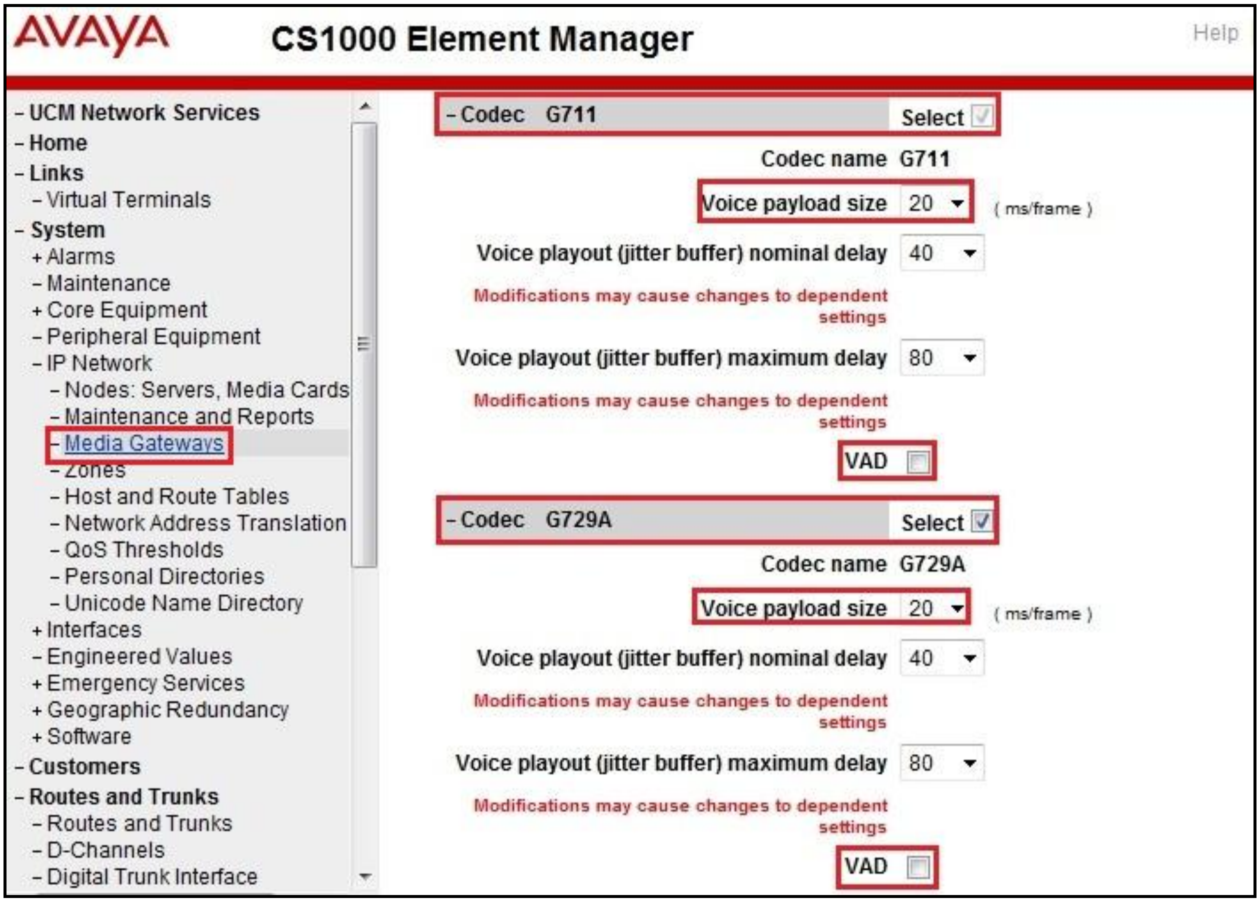


e) Click **Save**.
f) Synchronize the new configuration (refer to **Section 5.2.4** for more detail).

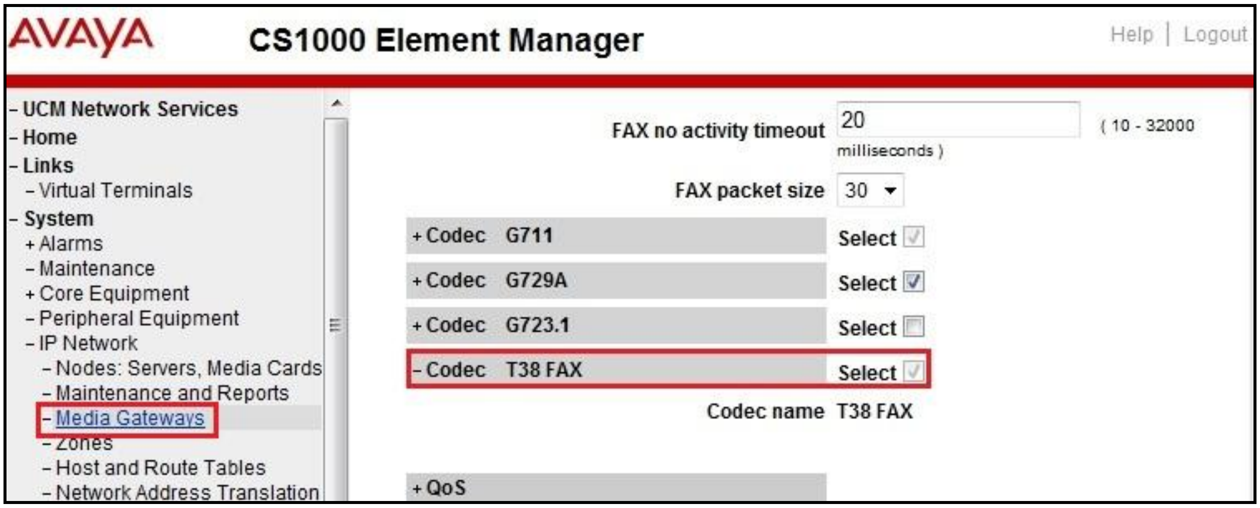
**Note**: Fax over IP using G.711MU codec is not supported, please refer to **Section 2.2**, observation #01 for detail information.

### 5.3.2. Administer Voice Codec on Media Gateways

The CS1000 uses media gateways to support traditional analog and digital phones calls over a SIP Trunk. Media gateways are also needed to support analog terminals and to send fax over IP.
a) To configure voice codecs for media gateways, from the left menu of the Element Manager page (not shown), select the **IP Network → Media Gateways** menu item. The Media Gateways page will appear (not shown). Click on **MGC** which is located on the right of the page (not shown).
b) Level 3 supports voice codecs G.729 and G.711, payload size 20 ms, with VAD disabled. The screenshot below shows appropriated codec profile configured for media gateways.




c) For Fax over IP, Level 3 supports T.38 codec. This parameter is enabled by default on CS1000 as shown in the following screenshot.




**Note**: Fax over IP using G.711MU codec is not supported, please refer to **Section 2.2**, observation #01 for detail information.

## 5.4. Administer Zones and Bandwidth

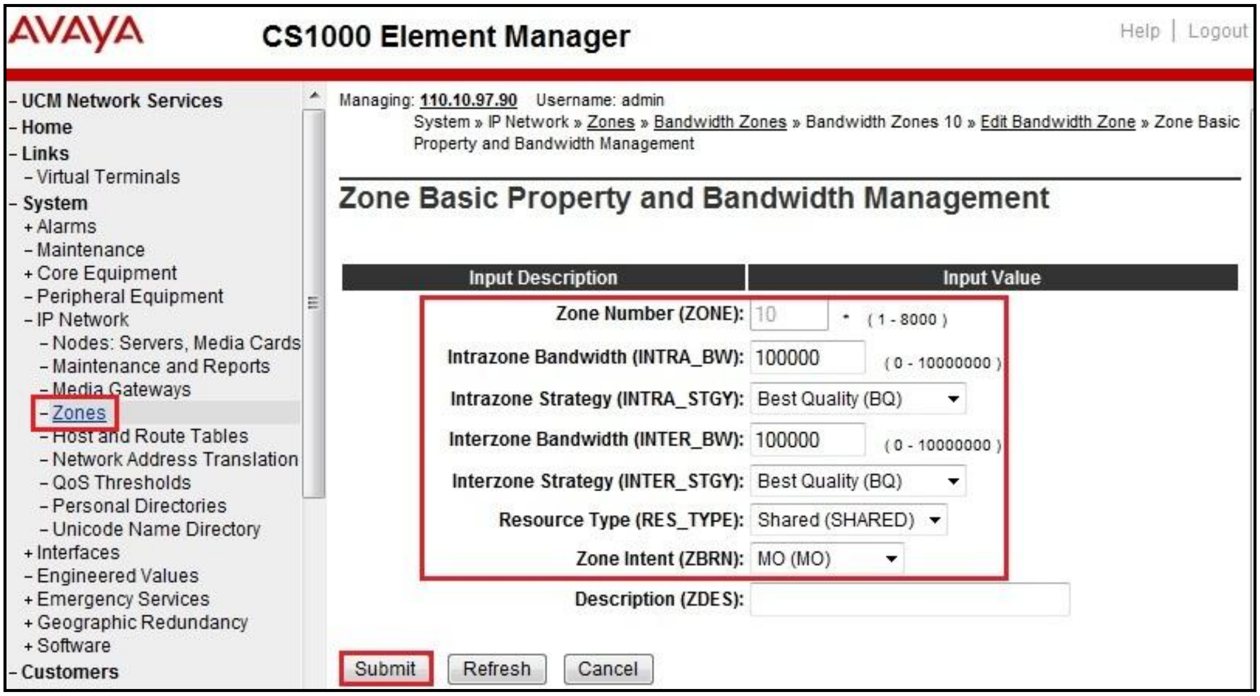
This section describes the steps to create 2 zones: zone 10 for Voice Gateway (VGW) and IP phones and zone 255 for a SIP Trunk. The CS1000 uses zone configuration for bandwidth management purposes.

### 5.4.1. Create a zone for IP phones

a) To create zone 10 for VGW and IP phone, select **IP Network → Zones** configuration from the left pane, click **Bandwidth Zones** link (not shown).
b) In **Bandwidth Zones** screen (not shown), click **Add** button (not shown).
c) In the **Add Bandwidth Zone** screen (not shown), click on **Zone Basic Property and Bandwidth Management**, select the values as shown (in red box) in the screenshot below and click on the **Submit** button.

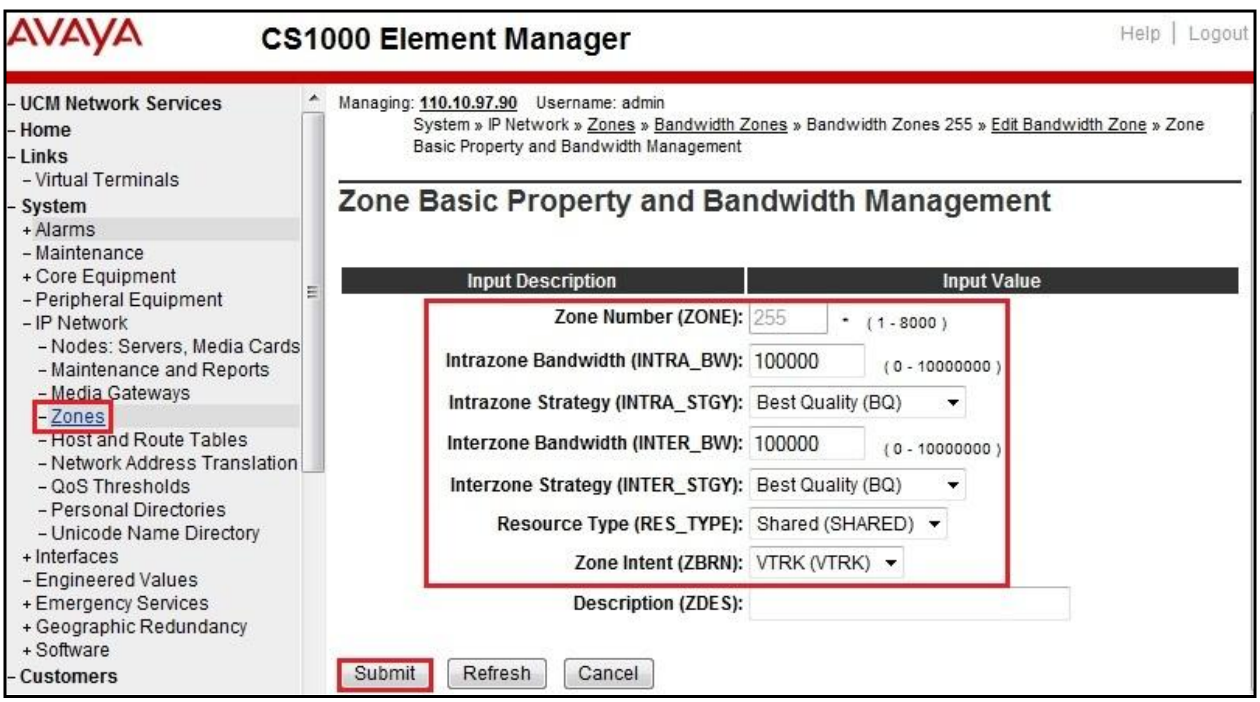
- **INTRA_STGY**: bandwidth configuration for local calls
- **INTER_STGY**: bandwidth configuration for the calls over trunk
- **BQ**: G.711 is first choice and G.729 is second choice
- **BB**: G.729 is first choice and G.711 is second choice
- **MO**: the zone type which is used for IP phones and Voice Gateway (VGW)
- **VTRK**: the zone type which is used for the SIP Trunk

Level 3 supports G.729 as the first choice, G.711. In the sample configuration as shown in the screenshot below, the **MO** Zone 10 is configured with **Strategy Best Quality (BQ)** to allow the CS1000 to select G.711MU as a first choice and G.729 as the second choice for a voice call.




## 5.4.2. Create a zone for virtual SIP trunk

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



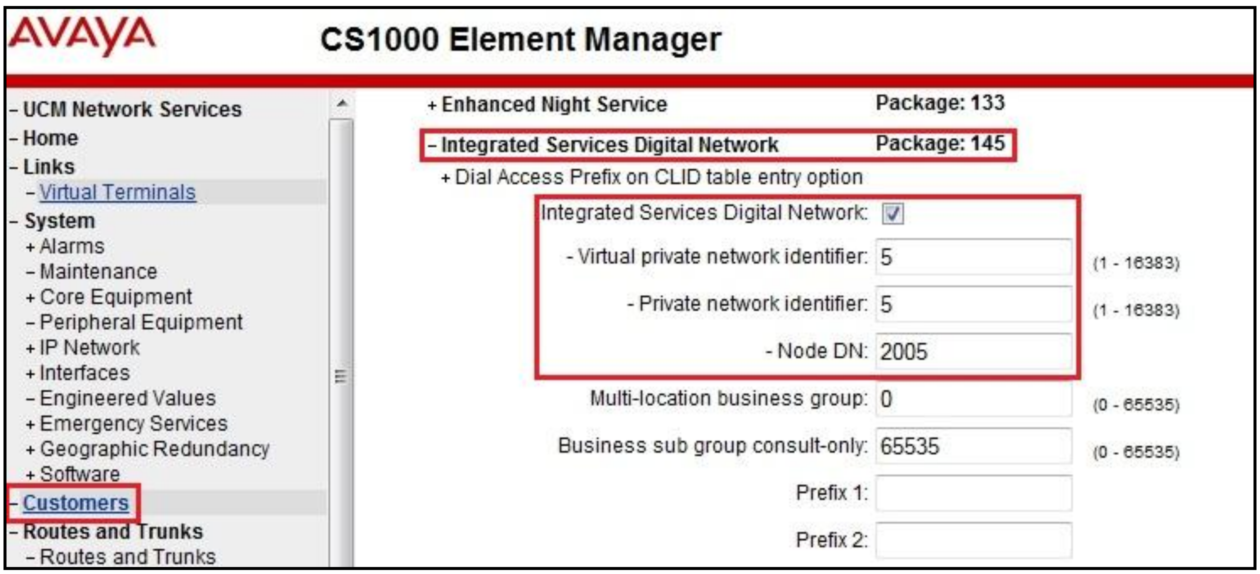
## 5.5. Administer SIP Trunk Gateway

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

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

a) To configure ISDN, select **Customers** in the left pane. The **Customers** screen is displayed (not shown). Click on the link associated with the appropriate customer, in this case is 05. The system can support more than one customer with different network settings and options. The **Customer 05 Edit** page will appear (not shown). Select the **Feature Packages** option from this page (not shown).

b) The screen is populated with a list of F**eature Packages**. Select **Integrated Services Digital Network** to edit its parameters. The screen expands with **Integrated Services Digital Network** parameters. Retain the default values for all remaining fields. Scroll down to the bottom of the screen, and click **Save** button (not shown)




### 5.5.2. Administer SIP Trunk Gateway to Session Manager

a) To configure SIP Trunk Gateway, select **IP Network → Nodes: Servers, Media Cards** configuration from the left pane, and in the **IP Telephony Nodes** screen, select the **Node ID** 2005. The **Node Details** screen is displayed as shown in **Section 5.2.1.**

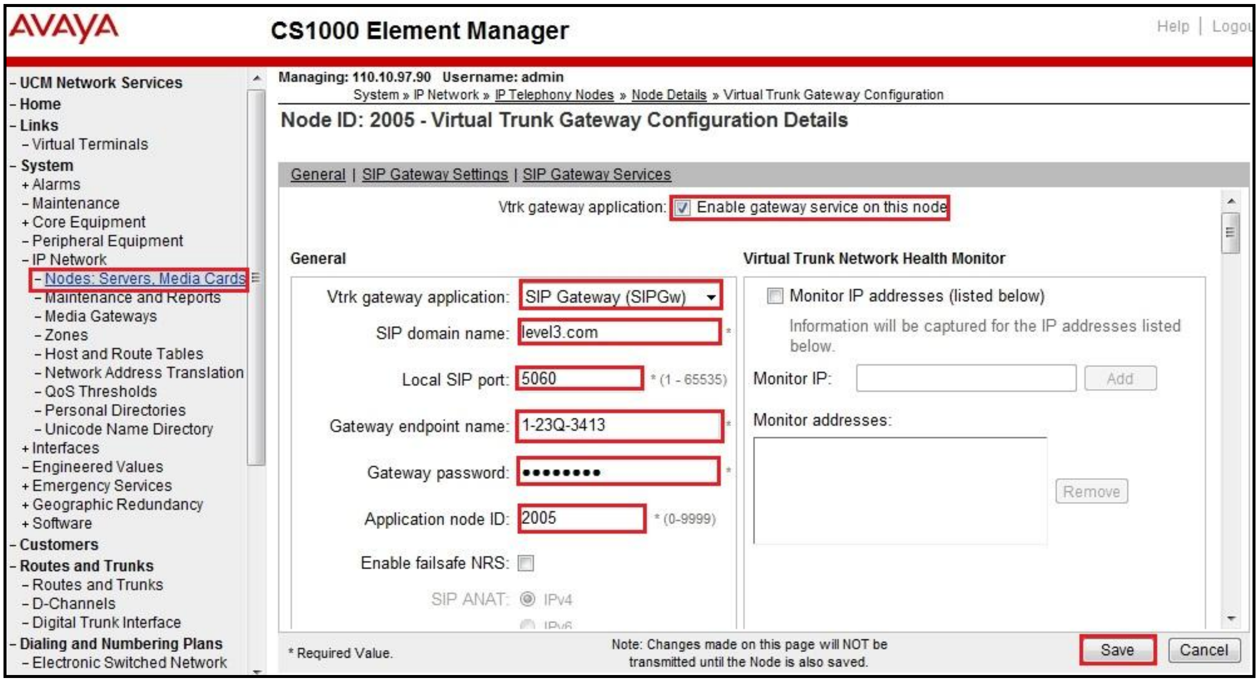
b) On the **Node Details** screen, select **Gateway (SIPGw)** (not shown).

c) Under **General** tab of the **Virtual Trunk Gateway Configuration Details** screen, enter the following values which are highlighted in red boxes as shown in screenshot below. These configurations are obtained when a user creates a SIP Entity on the Session Manager, these are shown in **Section 6.4**. Retain the default values for the remaining fields.

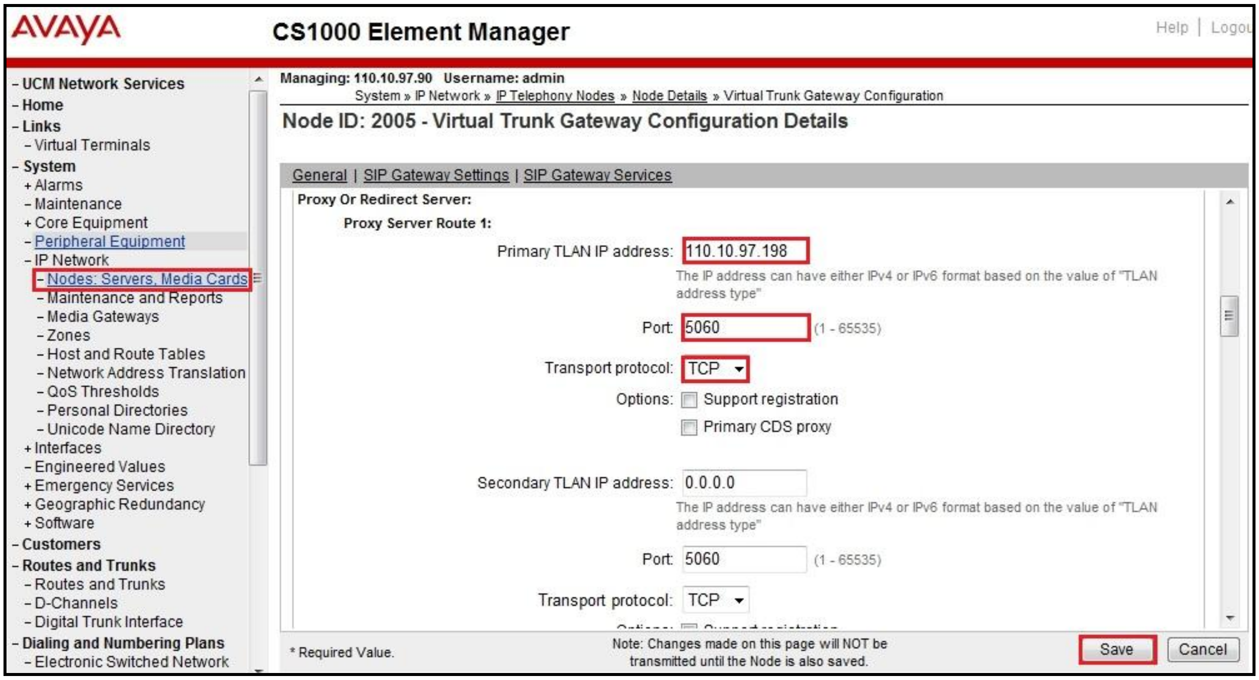
- **Vtrk gateway application**: SIP Gateway (SIPGw)
- **SIP domain name**: level3.com
- **Local SIP port**: 5060
- **Gateway endpoint name**: 1-23Q-3413 (This parameter is provided by Level 3)

- **Gateway password**: ****** (This parameter is provided by Level 3)
- **Application node ID**: 2005

**Note:** The gateway endpoint name and gateway password values are provided by Level 3, they are used by the CS1000 to construct a proper response to the Digest Authentication challenges implemented on the SIP Trunk by Level 3.




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



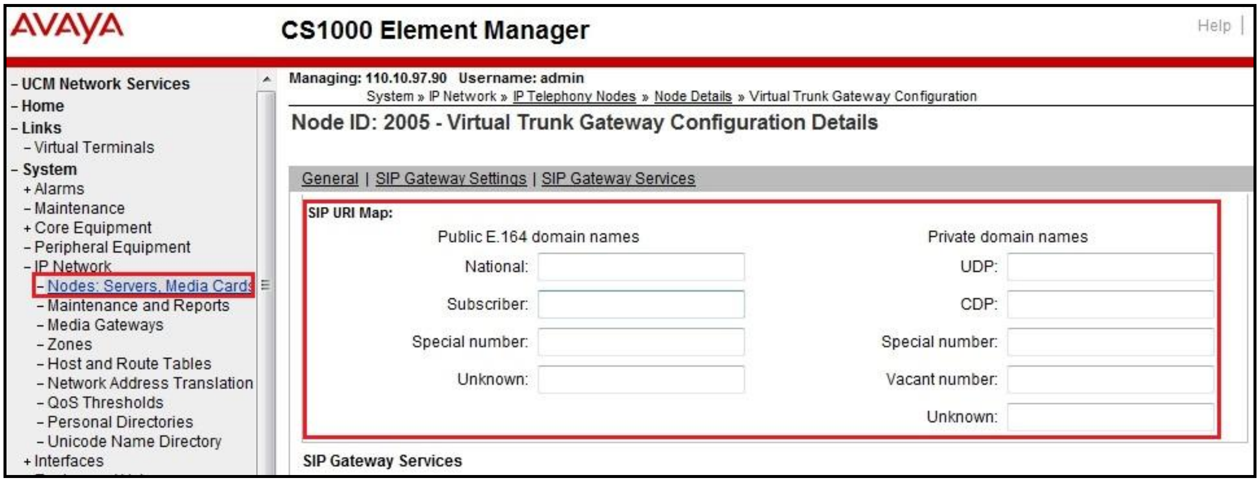

e) On the same page, scroll down to the **SIP URI Map** section as shown in the screenshot below. Under the **Public E.164 Domain Names**:

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

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

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

f) Then click **Save** button.
g) **Synchronize** the new configuration (refer to **Section 5.2.4**).

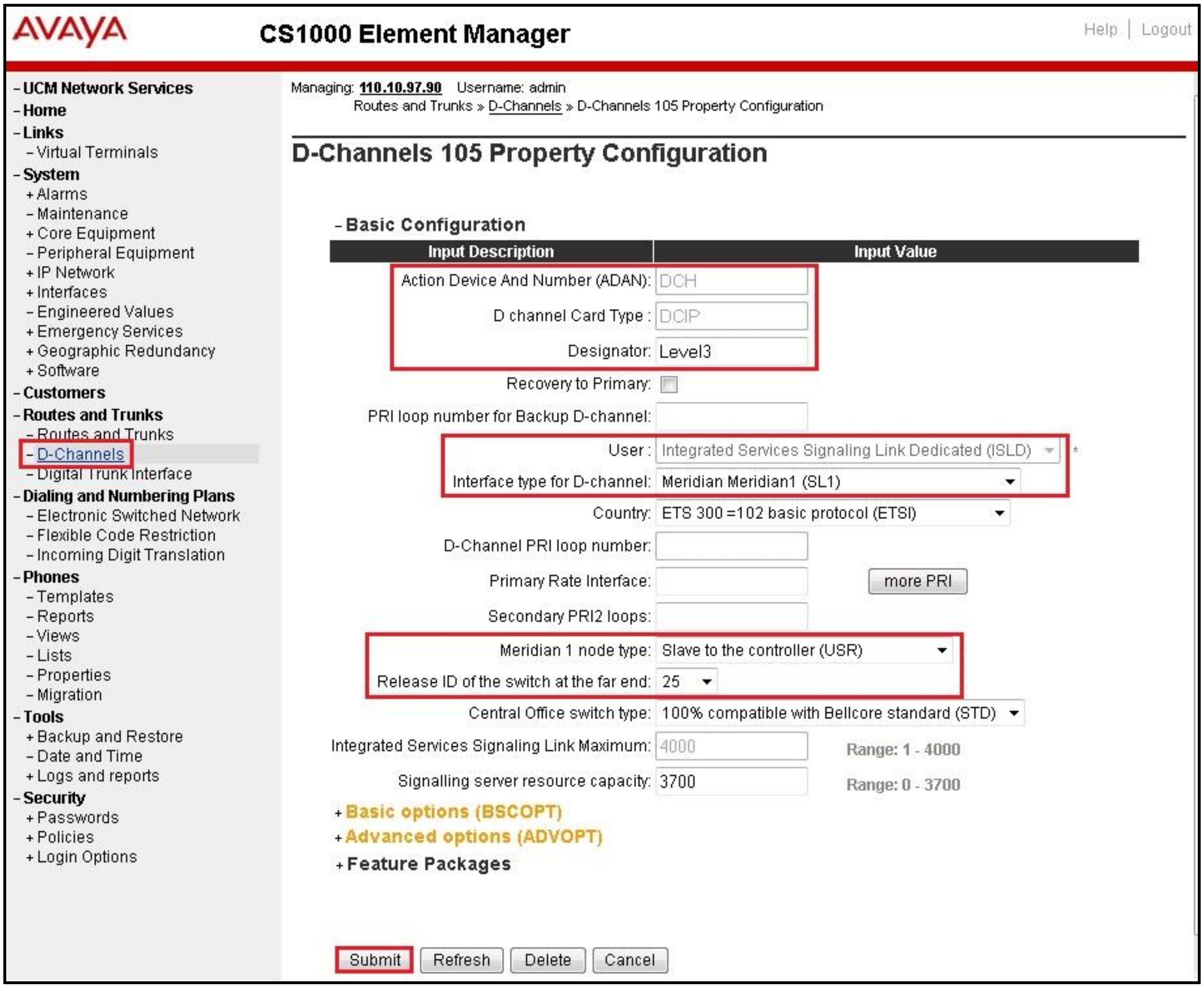



### 5.5.3. Administer Virtual D-Channel

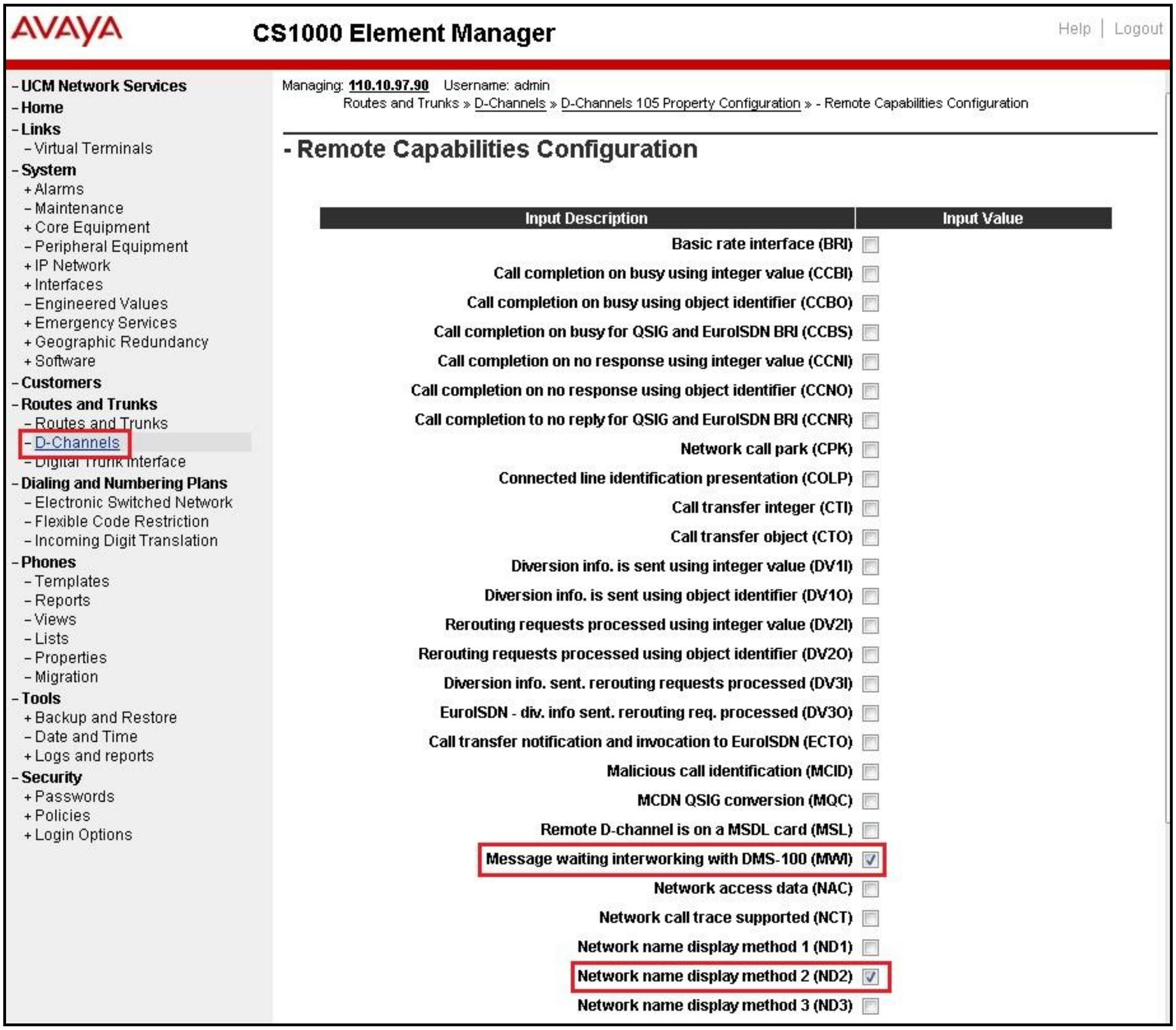
a) To create a D-Channel, select **Routes and Trunks → D-Channels** from the left pane to display the **D-Channels** screen (not shown). In the **Choose a D-Channel Number** field, select an available D-channel from the drop-down list (not shown). Click on **to Add** button (not shown)**.**

b) The **D-Channels Property Configuration** of DCH 105 is shown in the screenshot below. Enter the following values for the specified fields, and retain the default values for the remaining fields.

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

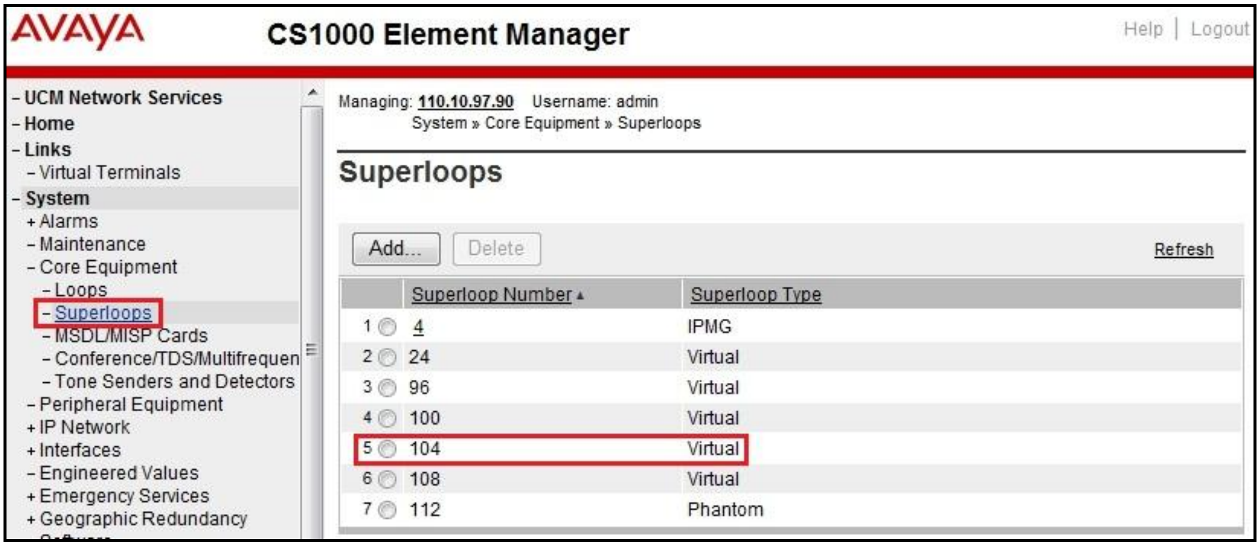



c) Continued from **D-Channels Property Configuration** described above, click on the **Basic Options** then click on the **Edit** button next to the **Remote Capabilities** (**RCAP**) attribute (not shown). The **Remote Capabilities Configuration** page will appear. Then check on the **ND2** and the **MWI** checkboxes as shown in the screenshot below.
d) Click **Return – Remote Capabilities** button (not shown).
e) Click **Submit** button (not shown).



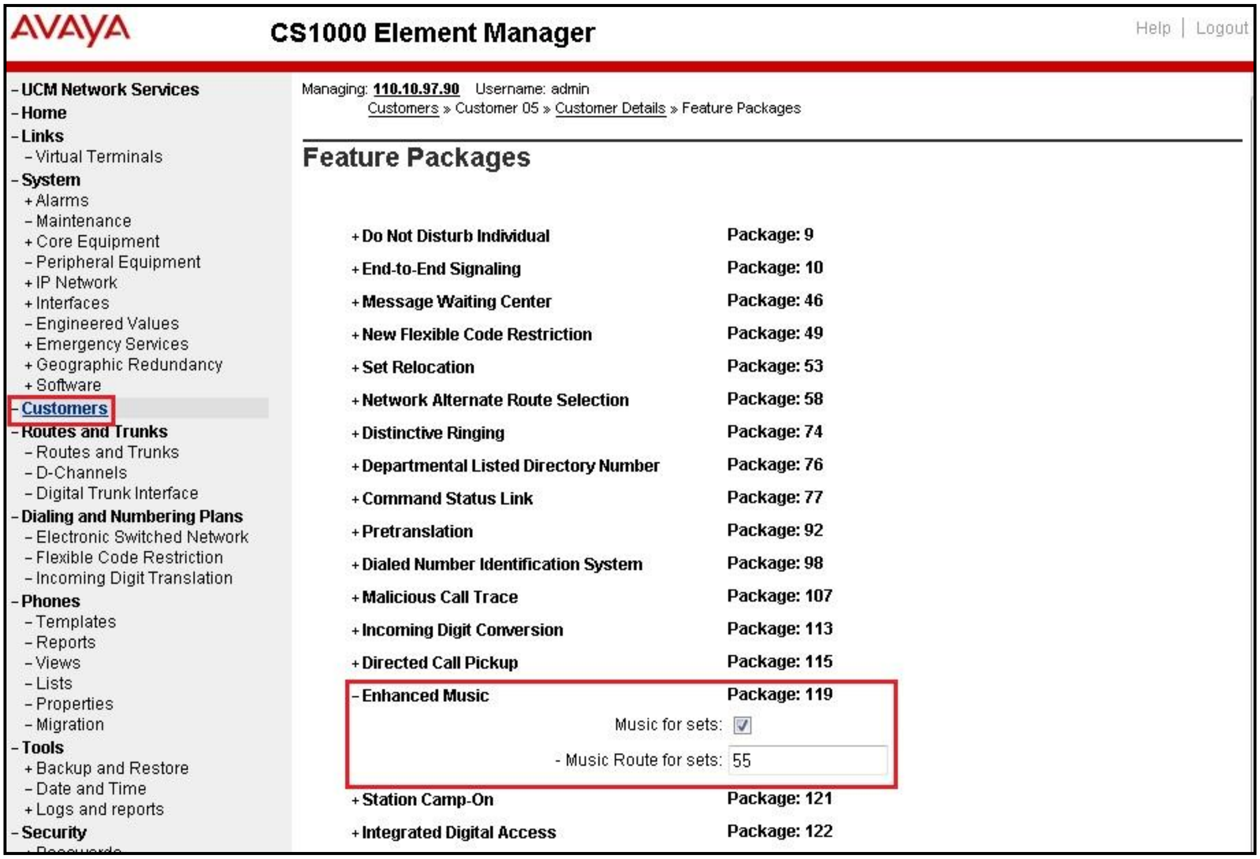

## 5.5.4. Administer Virtual Super-Loop

To add a virtual loop, select **System → Core Equipments → Superloops** from the left pane to display the **Superloops** screen. If the Superloop does not exist,then click "**Add**" button, provide an available virtual loop identification number then click the Save button (not shown)to create a new one as shown in the screenshot below. In this example, Superloop 104 is added.



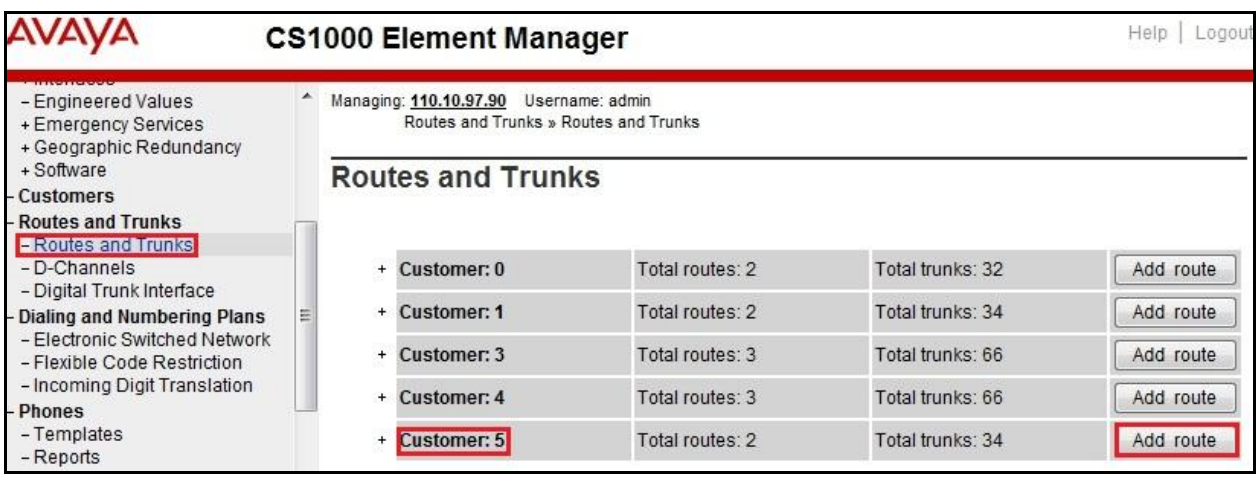

### 5.5.5. Enable Music for Customer Data Block

a) To enable music for a customer, select **Customers** in the left pane. The **Customers** screen is displayed (not shown). Click on the link associated with the appropriate customer, in this case is 05. The **Customer 05 Edit** page will appear (not shown). Select the **Feature Packages** option from this page (not shown).
b) The screen is populated with a list of F**eature Packages**. Select **Enhanced Music** to edit its parameters. Check to enable music for Customer 05, define music route 55 as shown in the red box of screenshot below. The CS1000 has been pre-configured with music route 55.



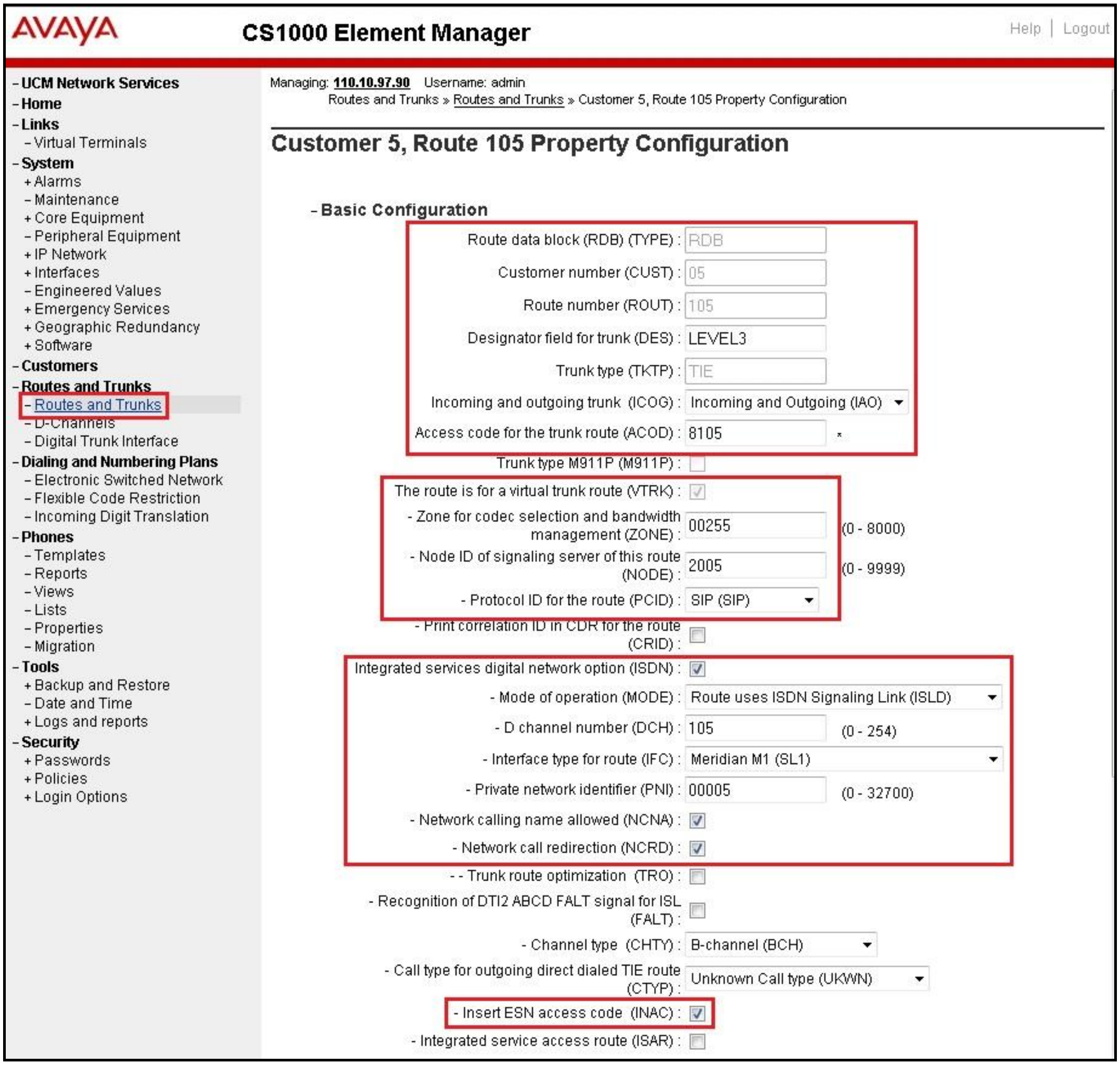

### 5.5.6. Administer Virtual SIP Routes

a) To create a SIP Route, select **Routes and Trunks → Routes and Trunks** from the left pane to display the **Routes and Trunks** screen. In this example, **Customer 05** is added. Click **Add route** button as shown in the screenshot below.

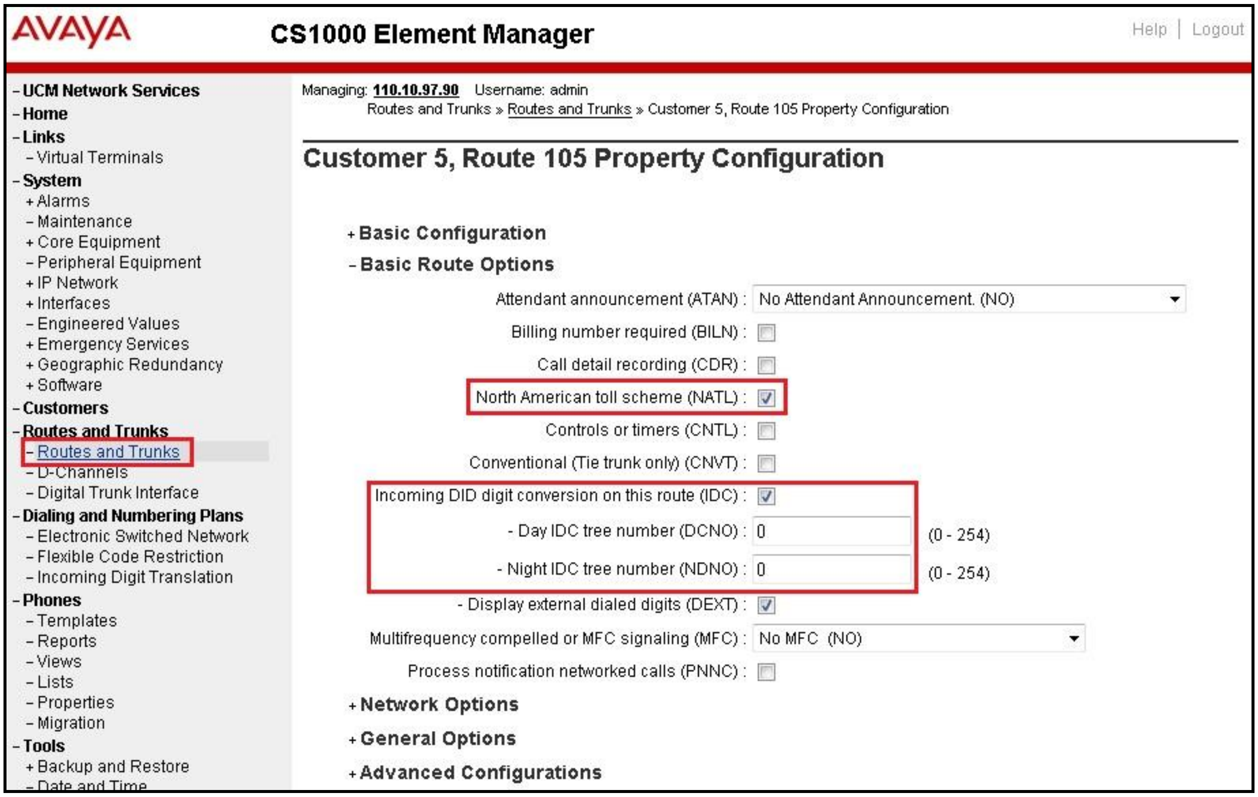



b) The **Customer 5**, New **Route Configuration** screen is displayed (not shown). Scroll down until the **Basic Configuration** section is displayed and enter the following values for the specified fields, and retain the default values for the remaining fields as shown in the screenshot below.

- **Route Number (ROUT)**: Select an available route number
- **Designator field for trunk (DES)**: A descriptive text
- **Trunk Type (TKTP)**: TIE trunk data block (TIE)
- **Incoming and Outgoing trunk (ICOG)**: Incoming and Outgoing (IAO)
- **Access Code for the trunk route (ACOD)**: An available access code
- Check the field **The route is for a virtual trunk route (VTRK**), to enable additional fields to appear
- For the **Zone for codec selection and bandwidth management (ZONE)** field, enter 255 (created in **Section 5.4.2**)
- For the **Node ID of signalling server of this route (NODE)** field, enter the node number 2005 (created in **Section 5.2.1**)
- Select **SIP (SIP)** from the drop-down list for **the Protocol ID for the route (PCID)** field
- Check the **Integrated Services Digital Network option (ISDN)** checkbox to enable additional fields to appear. Enter the following values for the specified fields, and retain the default values for the remaining fields
    - **Mode of operation (MODE)**: Route uses ISDN Signalling Link (ISLD)
    - **D channel number (DCH)**: D-Channel number 105 (created in **Section 5.5.3**)
    - **Network calling name allowed (NCNA)**: Checked
    - **Network call redirection (NCRD)**: Checked
    - **Insert ESN access code (INAC):** Checked

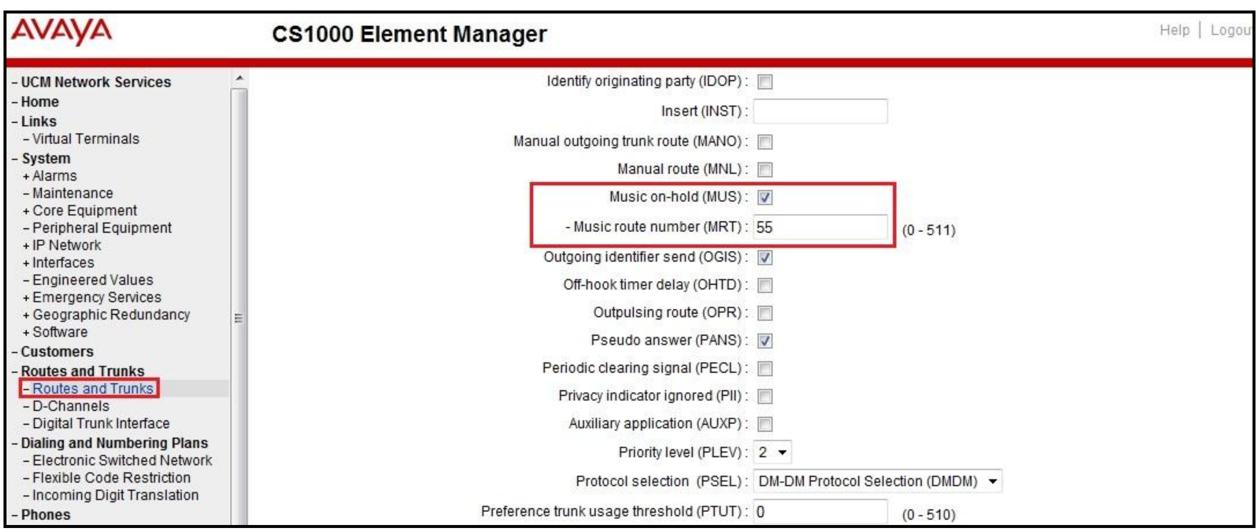



- Continued from **Route Configuration** described above, click on **Basic Route Options**, check **North American toll scheme (NATL)** and **Incoming DID digit conversion on this route (IDC)** and input DCNO 0 for both Day IDC Tree Number and Night IDC Tree Number as shown in screenshot below. The IDC is discussed in **Section 5.6.5**.



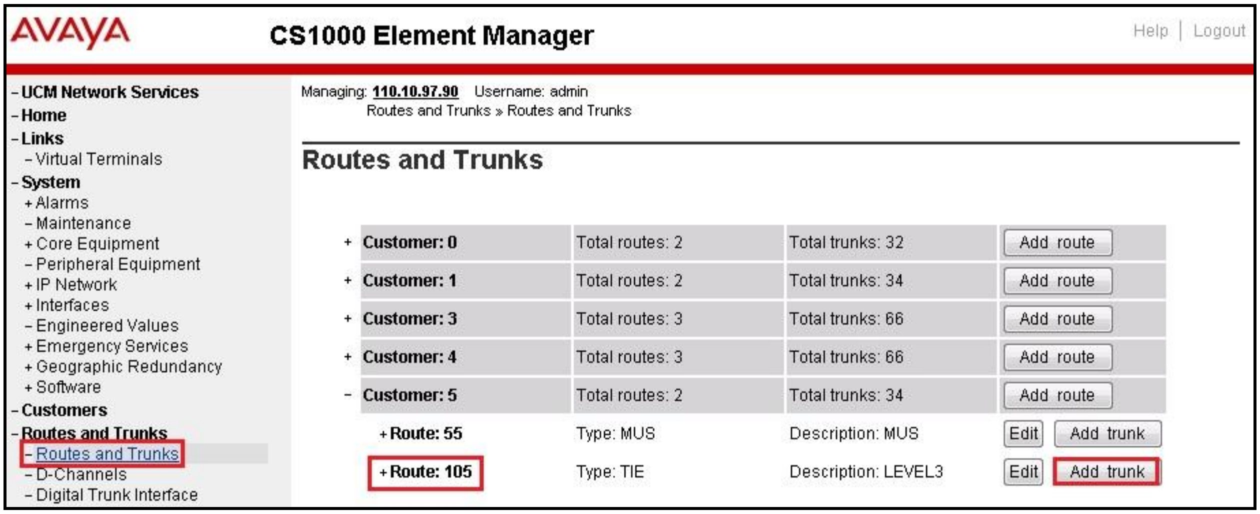

- Continued from **Route Configuration** described above, click on **Advance Configurations**; check **Music-on-holds** to enable music on hold on the route. Input music route 55 to the boxes as shown in the screenshot below. The CS1000 has been pre-configured with route 55 as a music route.

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



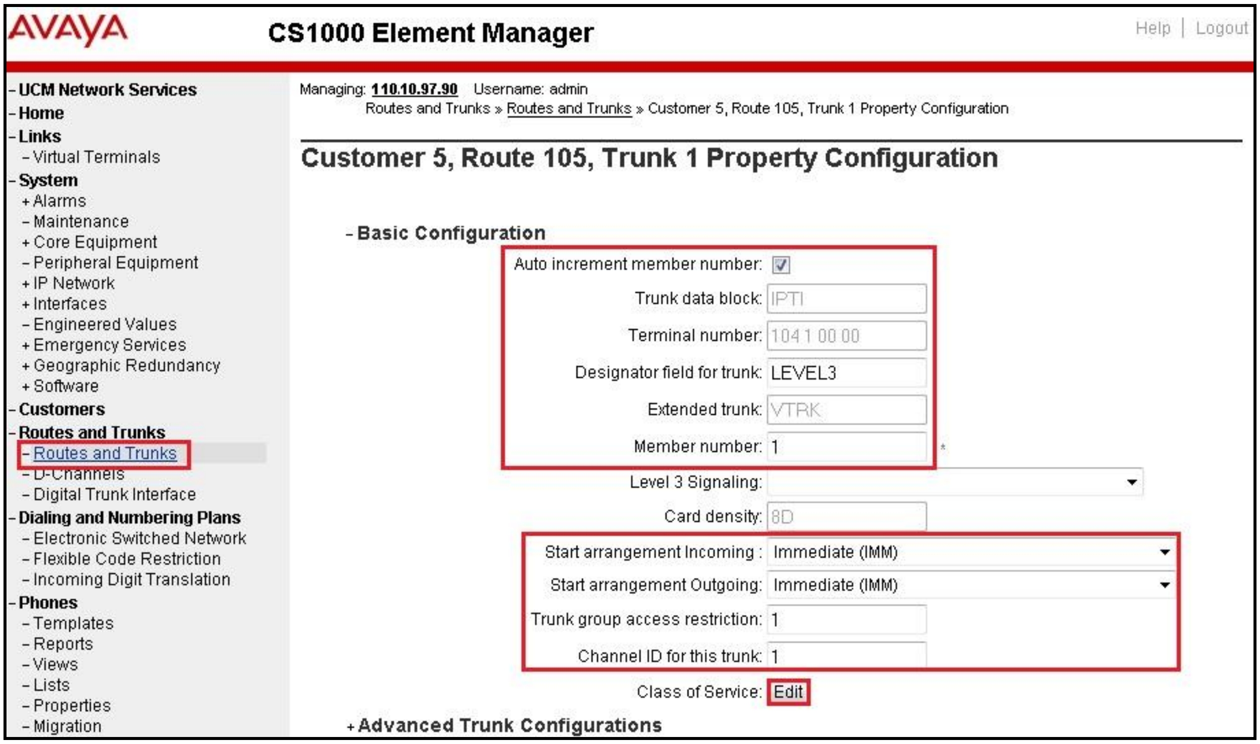
## 5.5.7. Administer Virtual Trunks

a) Continued from **Section 5.5.6**, the **Routes and Trunks** screen is displayed and updated with the newly added route (not shown). In the compliance test, route 105 is added. Click **Add trunk** button next to the newly added route 105 as shown in the screenshot below.

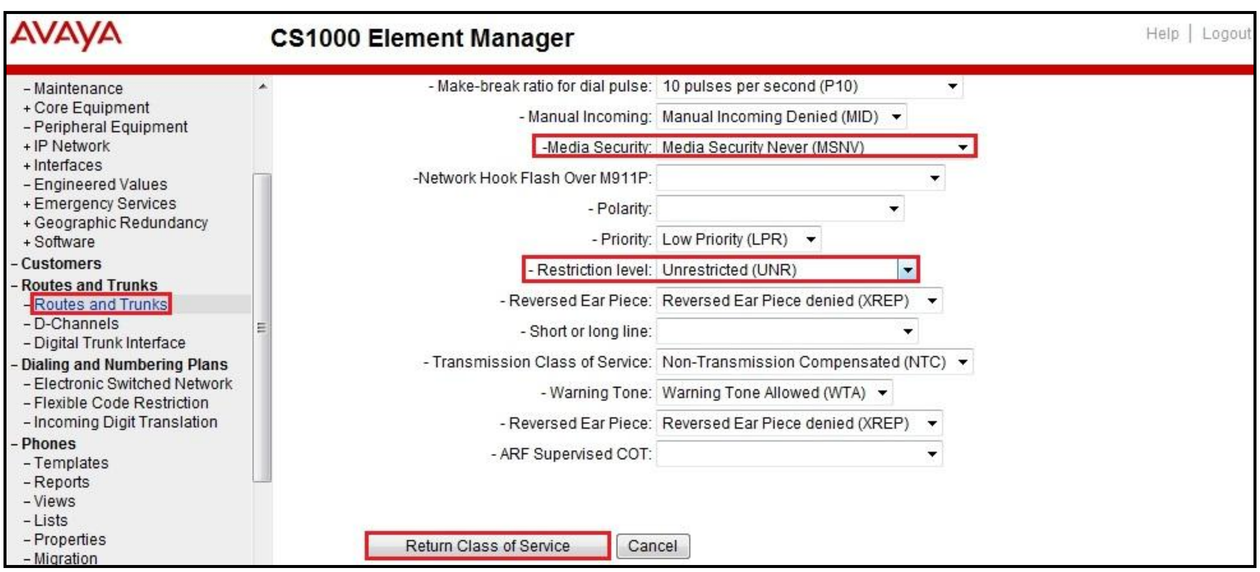



b) The **Customer 5, Route 105, Trunk 1 Property Configuration** is shown in the screenshot below. Enter **The Multiple trunk input number (MTINPUT)** field to add multiple trunks in a single operation, or repeat the operation for each trunk. In the certification test, 32 trunks are created (not shown). The following values are entered for specified fields and retain the default values for the remaining fields.

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




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



### 5.5.8. Administer Calling Line Identification Entries

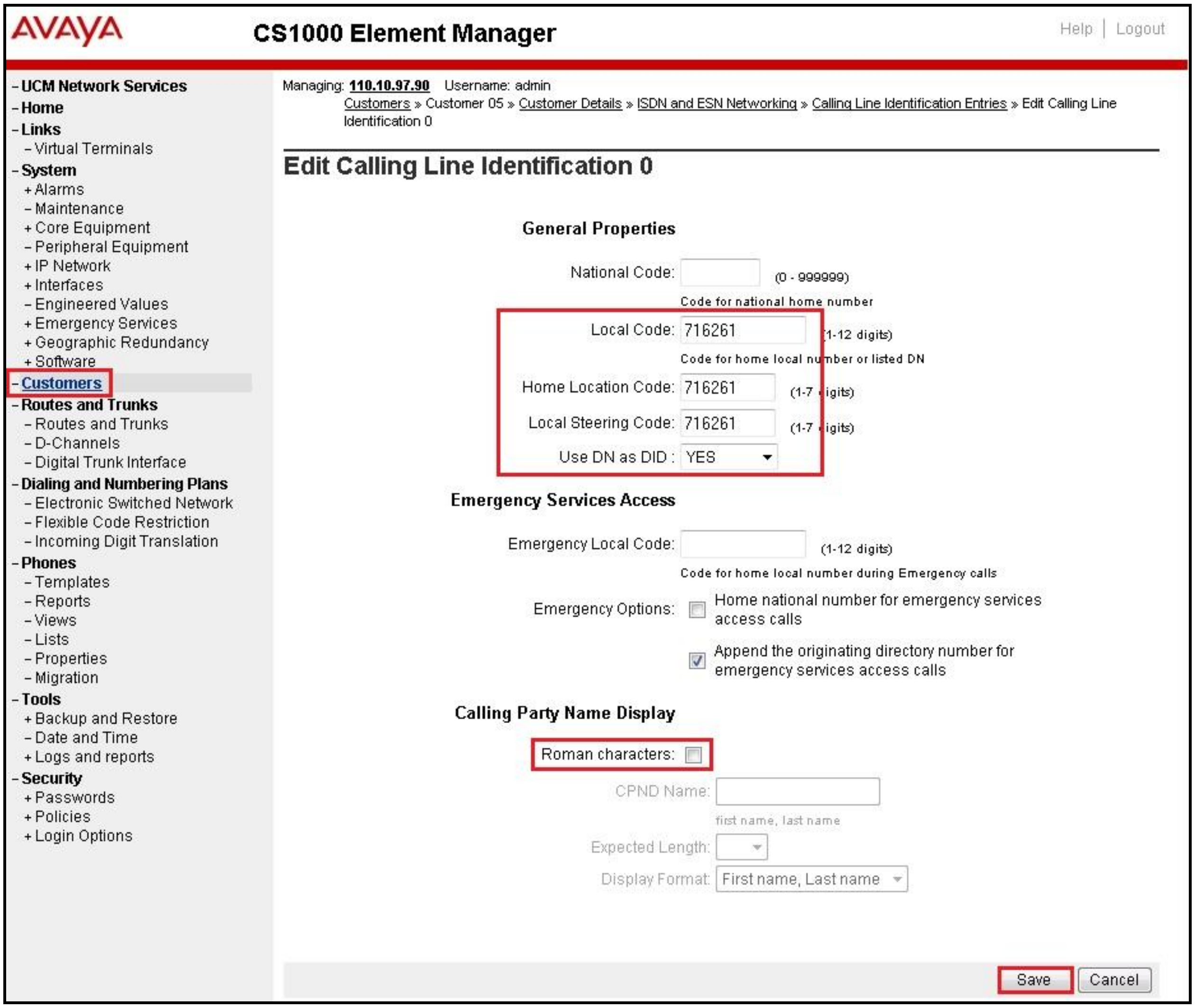
a) To create a Calling Line Identification Entry, select **Customers** > **05** > **ISDN and ESN Networking**. Click on **Calling Line Identification Entries** link at the bottom of the page (not shown)

b) On the Calling Line Identification Entries page (not shown), click **Add**.

c) Add entry **0** as shown in the screenshot below**.**

- **National Code**: leave as blank
- **Local Code:** input prefix digits assigned by Service Provider, in this case it is 6 digits – 716261. This **Local Code** is used for call display purpose of outbound international call configuration in **Section 5.6.6** where the Special Number 0 is associated with Call Type = Unknown
- **Home Location Code**: input prefix digits assigned by Service Provider, in this case it is 6 digits - 716261. This **Home Location Code** is used for call display purpose for Call Type = National (NPA)
- **Local Steering Code**: input prefix digits assigned by Service Provider, in this case it is 6 digits - 716261. This **Local Steering Code** is be used for call display purpose for Call Type = Local Subscriber (NXX)
- **Calling Party Name Display**: Uncheck Roman characters

d) Click **Save** button.




## 5.5.9. Enable External Trunk to Trunk Transferring

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

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

b) Allow **External Trunk To Trunk Transferring** for **Customer Data Block** by using LD 15.

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

TYPE NET_DATA
CUST 5
OPT
…
TRNX YES
EXTT YES
```

```
…
```

### 5.5.10. Enable plug-in 501

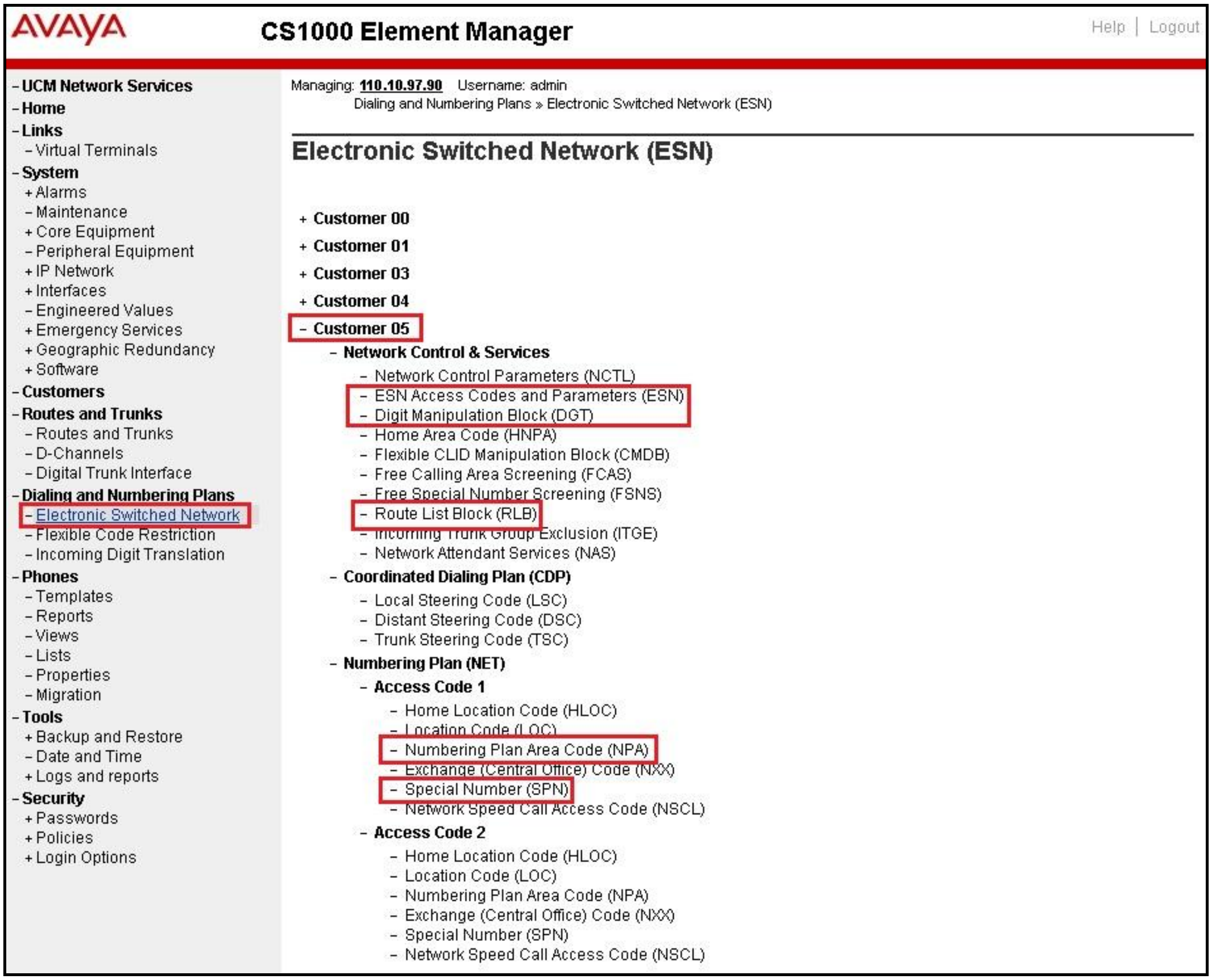
This section shows how to enable plug-in 501 in pdt mode to support blind transfer without UPDATE method. For more information, please refer to **Section 2.2**, observation #02.
a) Login Call Server CLI (please refer to **Section 5.1.2** for more detail).
b) Press Ctrl + pdt.
c) Login using user name as admin and provide proper password.
d) Issue command "ple 501" to enable plug-in 501.

```
PDT login on /pty/ptty01.S
Username: admin
Password:

The software and data stored on this system are the property of,
or licensed to, Avaya Inc. and are lawfully available only to
authorized users for approved purposes. Unauthorized access to
any software or data on this system is strictly prohibited and
punishable under appropriate laws. If you are not an authorized
user then logout immediately. This system may be monitored for
operational purposes at any time.
pdt> ple 501

PLUG-IN 501  IS ENABLED

pdt>
```

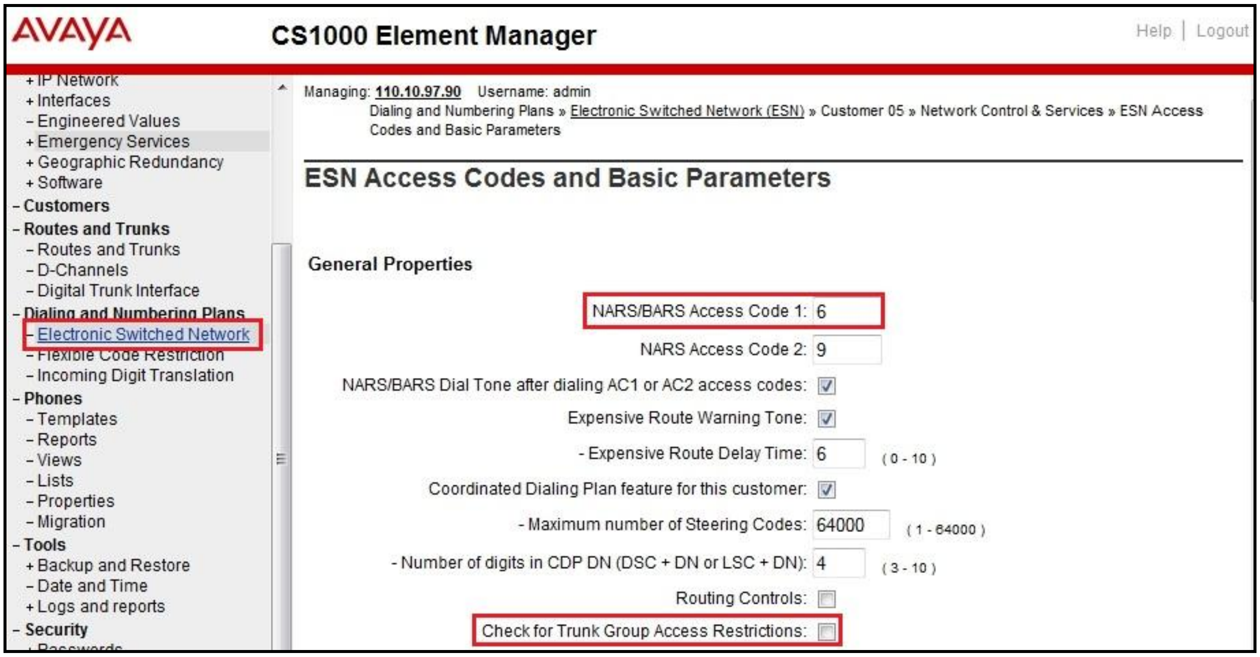
## 5.6. Administer Dialing Plans

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

a) Select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (**ESN**) screen. When Administering Dial Plans, the highlighted sections below will be configured in this section in the order they appear on the screen. To configure ESN parameter, select **ESN Access Code and Parameters (ESN)**.




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




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

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

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

TYPE NET_DATA
CUST 5
OPT
AC2 xNPA xSPN
FNP
CLID
…
```

c) Verify Customer Net_Data block by using LD 21.

```
>ld 21
PT1000

REQ: prt
TYPE: net
TYPE NET_DATA
CUST 4

TYPE NET_DATA
CUST 01
```

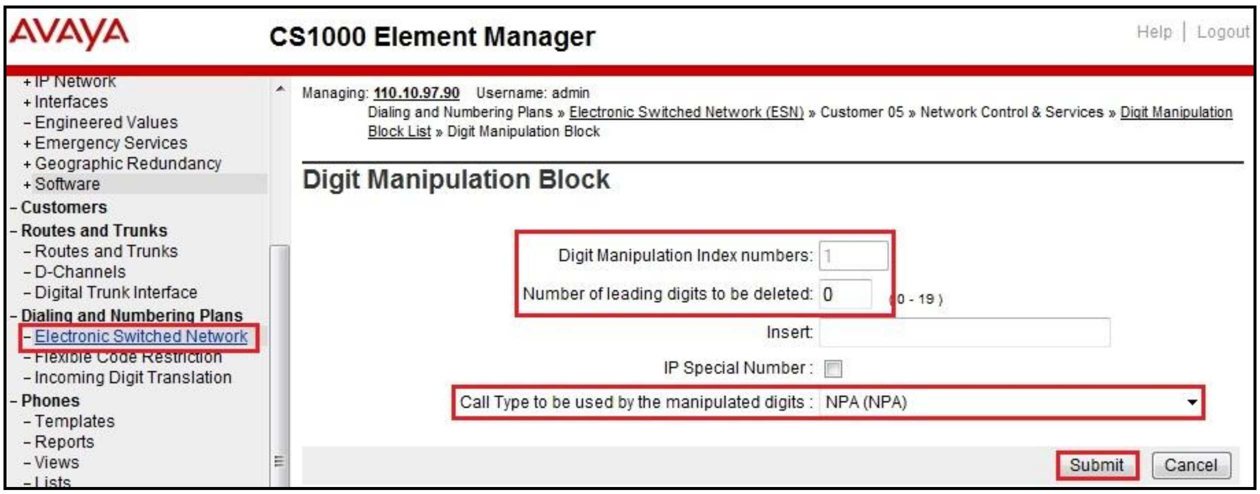
```
OPT RTA
AC1 INTL NPA SPN NXX LOC
AC2
FNP YES
…
```

### 5.6.3. Digit Manipulation Block (DMI)

a) To create a DMI, select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen (not shown).
b) Select **Digit Manipulation Block** (DGT) as shown in **Section 5.6.1**.b) In the **Choose a DMI Number** field, select an available DMI from the drop-down list and click **to Add** (not shown).
c) The screeshot below shows DMI 1 is created with following values.

- **Number of leading digits to be Deleted** (Del): 0
- **Call Type to be used by the manipulated digits** (CTYP): NPA

d) Click **Submit** button.

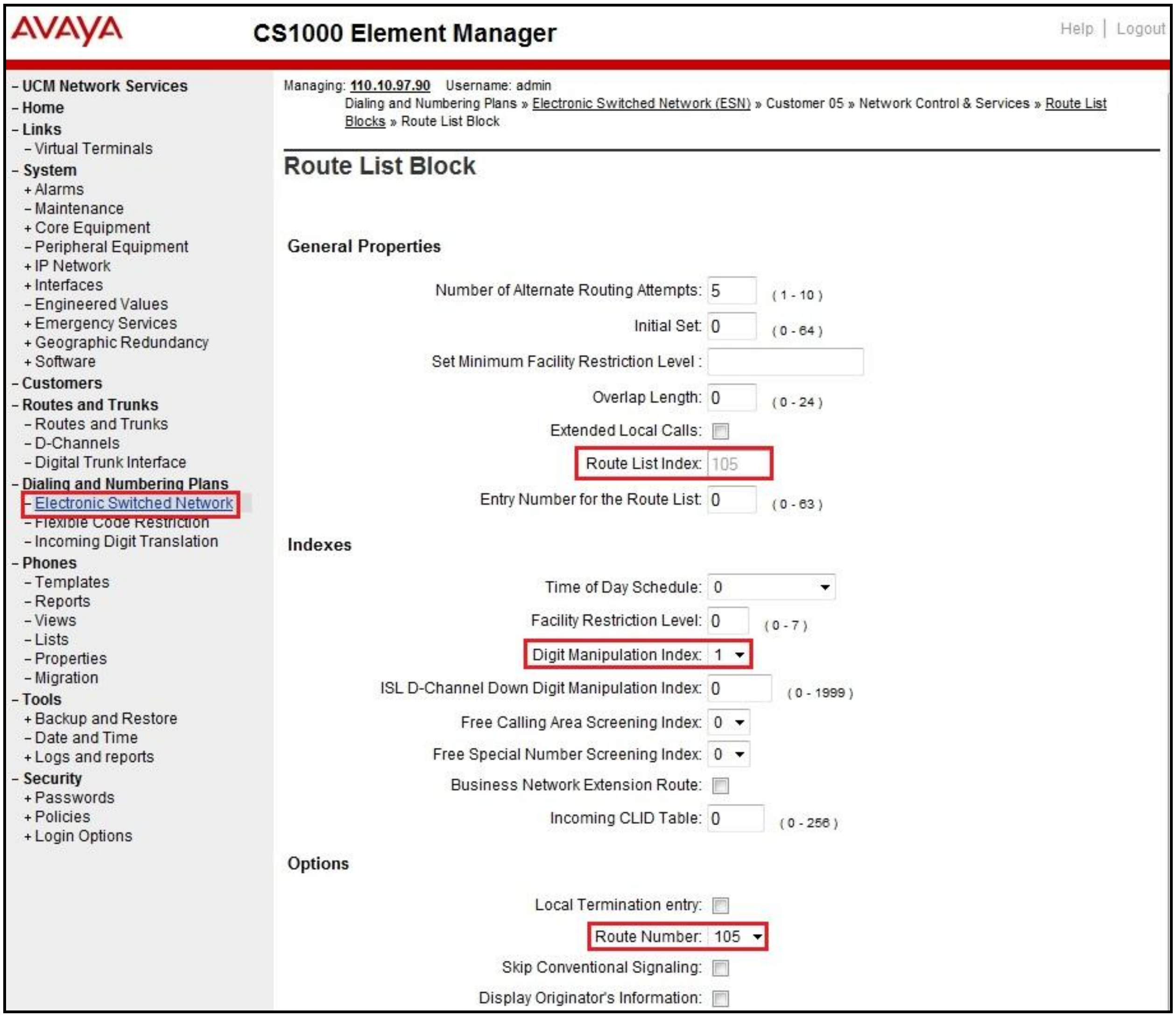



### 5.6.4. Route List Block (RLB)

This section shows how to add a RLB associated with the DMI created in **Section 5.6.3**.
a) To create RLB 105, select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen. Select **Route List Block** (RLB) as shown in **Section 5.6.1**.
b) Select an available value .e.g. 105 in the textbox for the **route list index** and click on the "**to Add"** button (not shown).
c) Enter the following values for the specified fields, and retain the default values for the remaining fields as shown in the screenshot below.

- **Route number (ROUT)**: 105 (created in **Section 5.5.6**)
- **Digit Manipulation Index (DMI)**: 1 (created in **Section 5.6.3**)

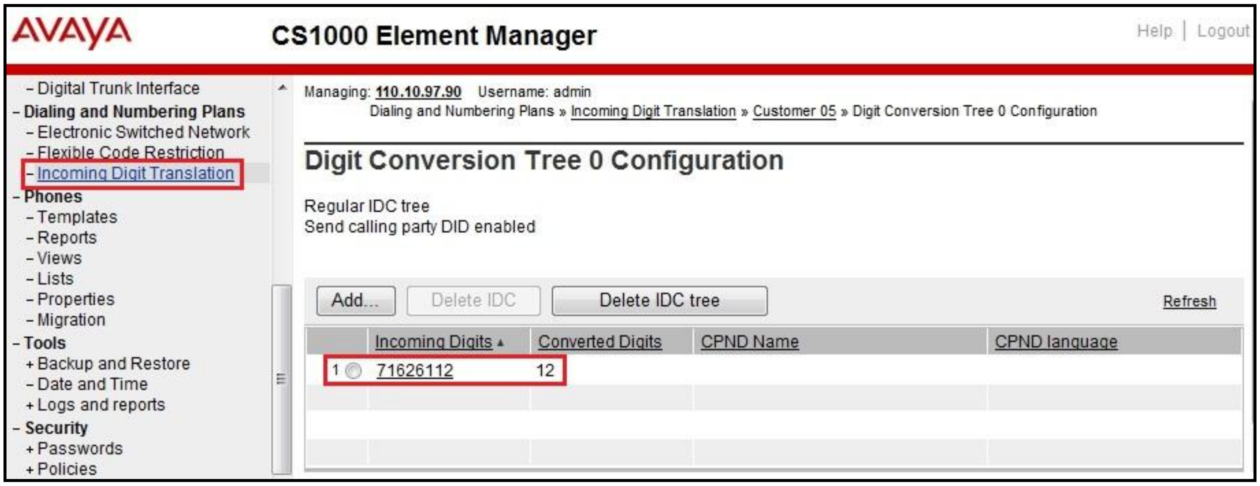
d) On the same page, scroll down to the bottom of the screen, and click **Submit** button (not shown).




## 5.6.5. Incoming Digit Translation (IDC)

This section describes the steps for receiving calls from the PSTN via Level 3.
a) To create an IDC, 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 (not shown).
b) Click on **New DCNO** to create a digit translation entry. In this example, Digit Conversion Tree Number (**DCN0**) **0** is created. Detail configuration of the **DCNO** is shown in screenshot below. The **Incoming Digits** can be added to map to the **Converted Digits** which would be the CS1000 DN.  This **DCN0** has been assigned to route 105 as shown in **Section 5.5.6**.

In the following configuration, incoming calls from PSTN with prefix 71626112XX will be translated to CS1K DN 12XX, including the DID 7162611214 is translated to 1214 for voice mail access purpose.




### 5.6.6. Outbound Call - Special Number Configuration

Special numbers are configured for this testing. For example, 0 to reach an operator, 0+10 digits to reach operator assistant, 011 prefix for international calls, 1 for a national long distance call, 411 for directory assistant and so on.

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

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

Special Number: 0

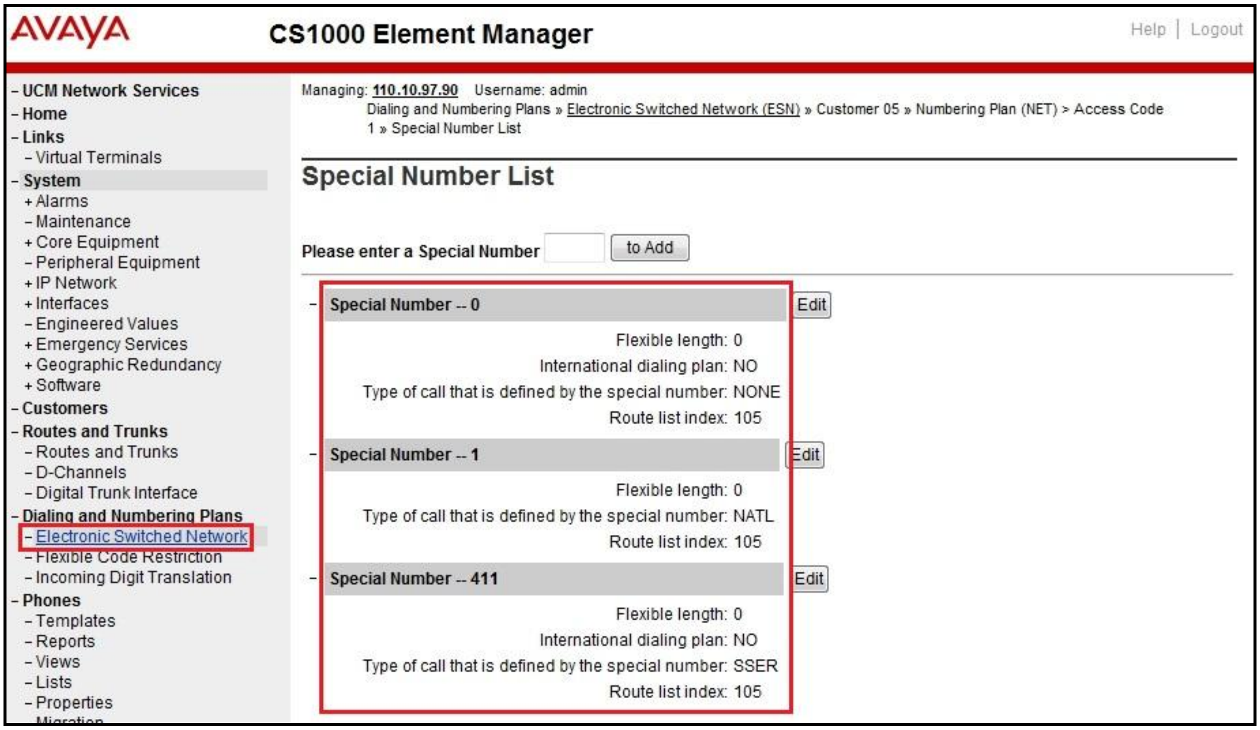
- **Flexible length**: 0 (flexible, unlimited and accept the character # to ending dial number).
- **Call Type**: NONE.
- **Route list index**: 105, created in **Section 5.6.4**.

Special Number: 1

- **Flexible length**: 0 (flexible, unlimited and accept the character # to ending dial number).
- **Call Type**: NATL.
- **Route list index**: 105, created in **Section 5.6.4**.

Special Number: 411

- **Flexible length**: 3.
- **CallType**: SSER.
- **Route list index**: 105, created in **Section 5.6.4**.

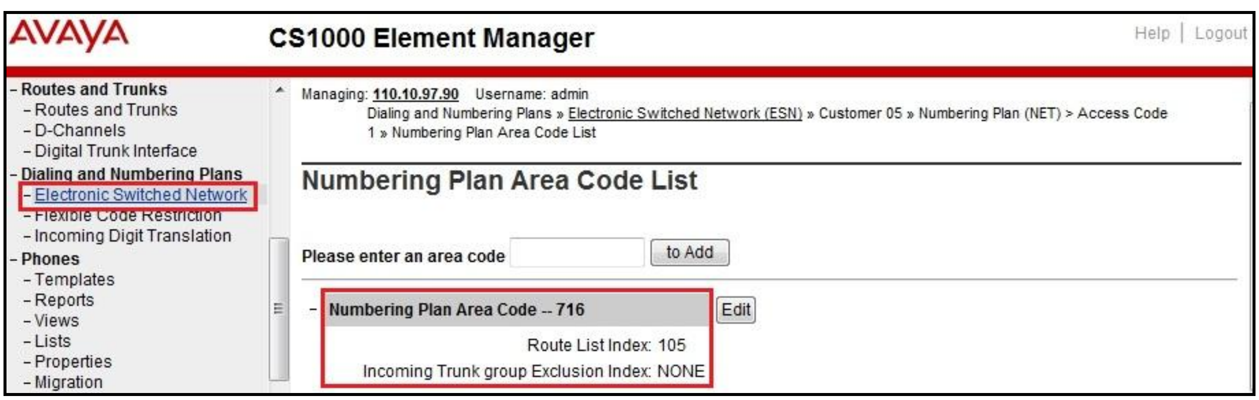



### 5.6.7. Outbound Call - Numbering Plan Area (NPA)

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

a) To create a NPA number, select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen (not shown). Select **Numbering Plan Area Code** (NPA) (not shown).

b) Enter area code desired in the textbox and click "**to Add"** button (not shown). The screenshot below shows NPA numbers 716 is configured for this testing. These NPA numbers are associated to the SIP Trunk for 10-digit outbound local call.




# 6. Configure Avaya Aura® Session Manager

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

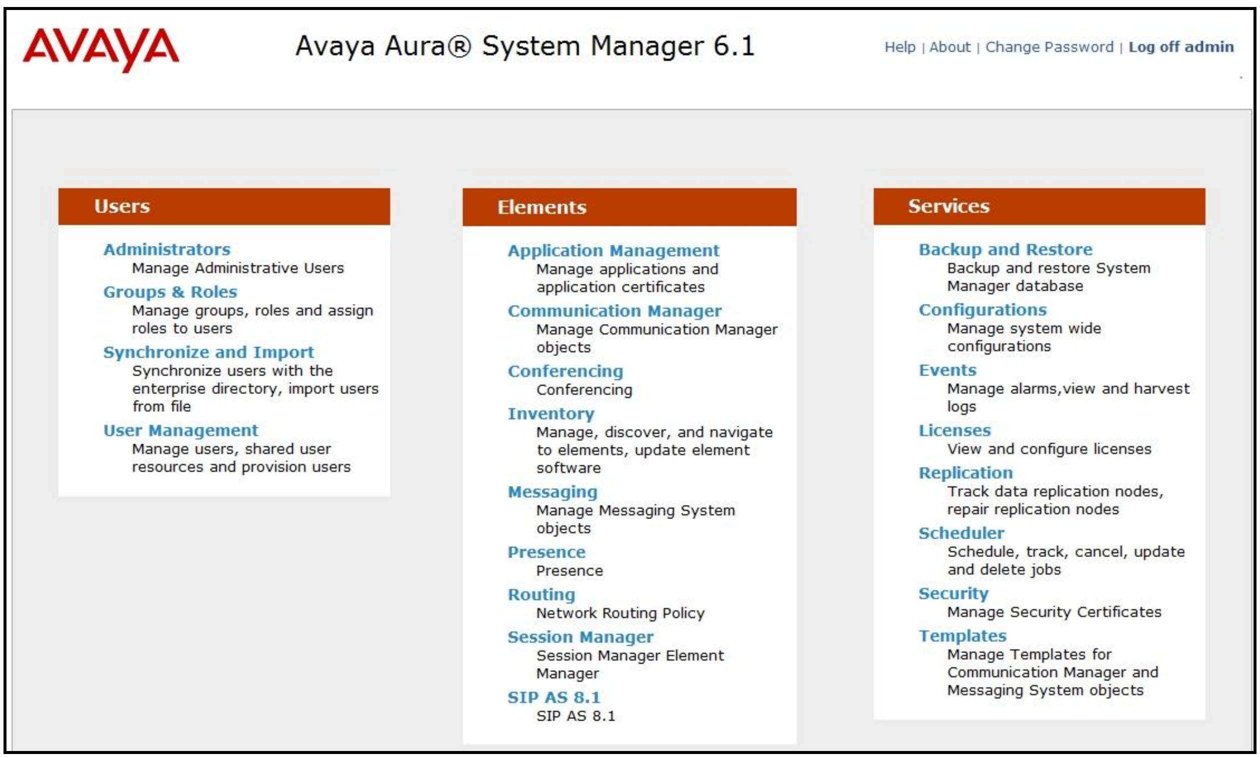
- SIP domain

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

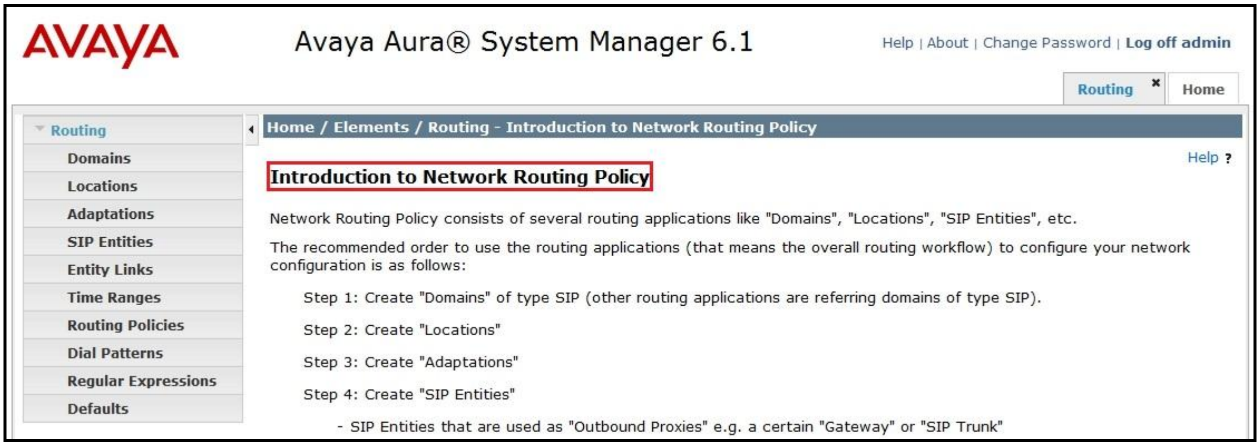
It may not be necessary to create all the items above when creating a connection to the service provider since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP domains, locations, SIP entities, and Session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

## 6.1. System Manager Login and Navigation

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




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

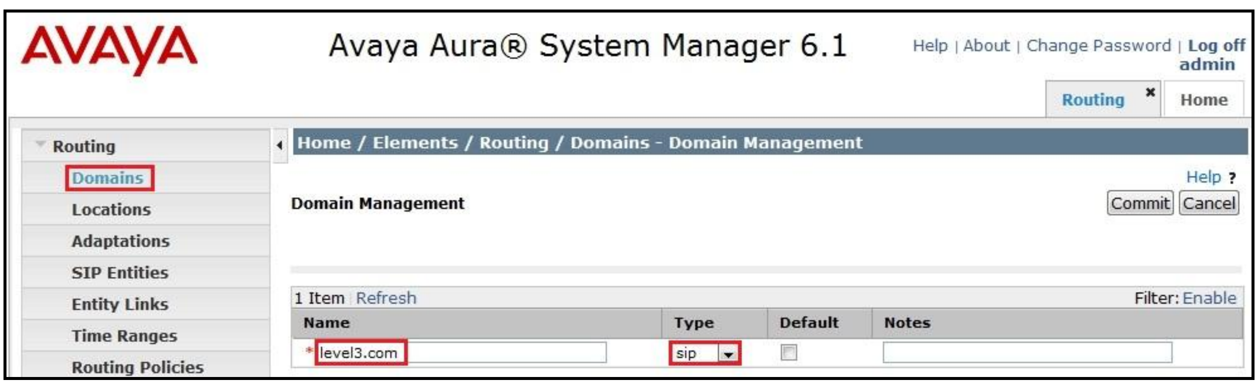



The navigation tree displayed in the left pane will be referenced in subsequent sections to navigate to items requiring configuration.

## 6.2. Specify SIP Domain

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

The following screenshot shows domain **level3.com** is already created. It is used for communication among a number of Avaya systems and applications with SIP integration to Session Manager. The domain **level3.com** is not known to Level 3. Later, it will be adapted by the Acme SBC to an IP address based URI-Host to meet the requirements of Level 3.




## 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.

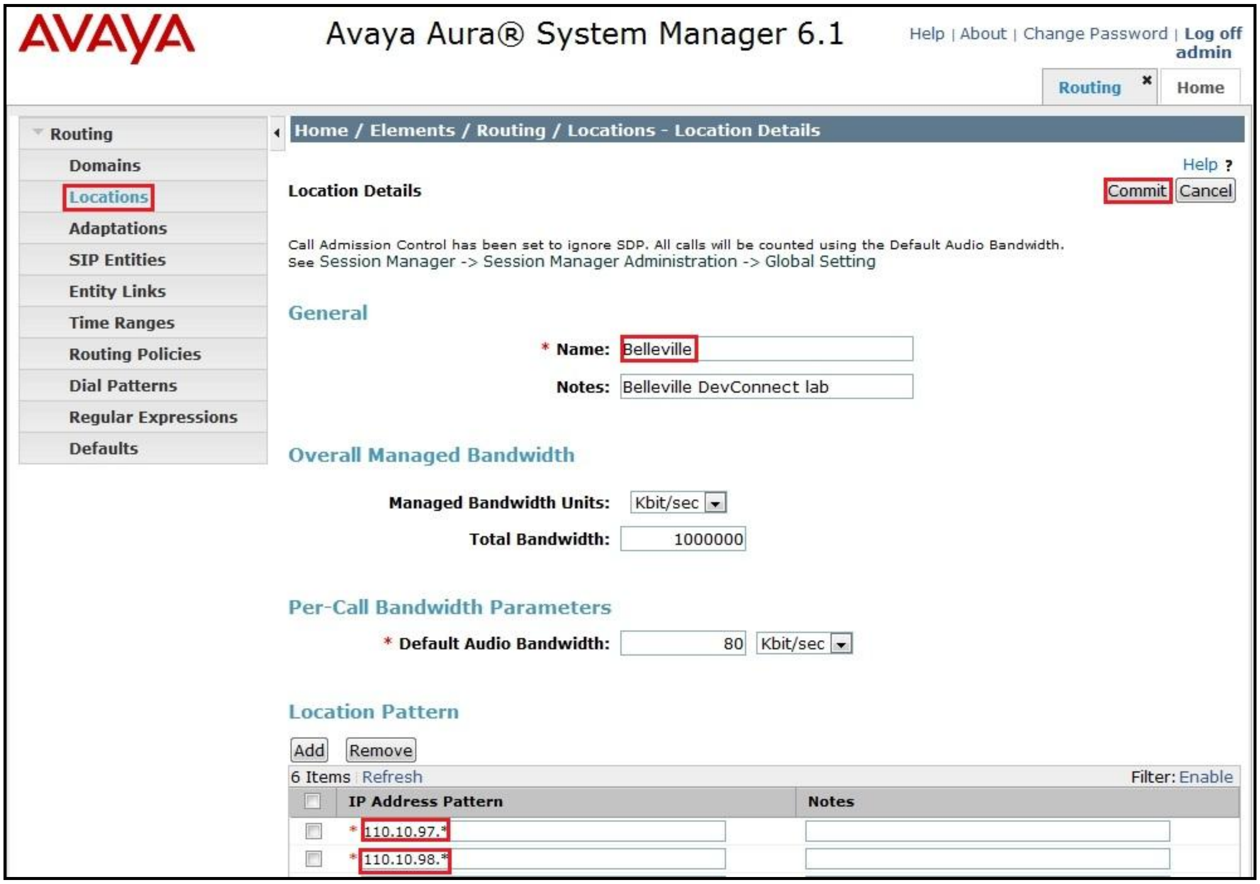
To add a location, navigate to **Routing →Locations** in the left-hand navigation pane and click the **New** button in the right pane (not shown).

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

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

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

- **IP Address Pattern**: Enter two subnets 110.10.97.x and 110.10.98.x which are IP address patterns used to identify the location including the CS1000, Session Manager and the Acme SBC
- **Notes:** Add a brief description (optional)
- Click **Commit** button.




## 6.4. Add SIP Entities

A SIP Entity must be added for Session Manager and for each SIP telephony system connected to it which includes CS1000 and Acme SBC.

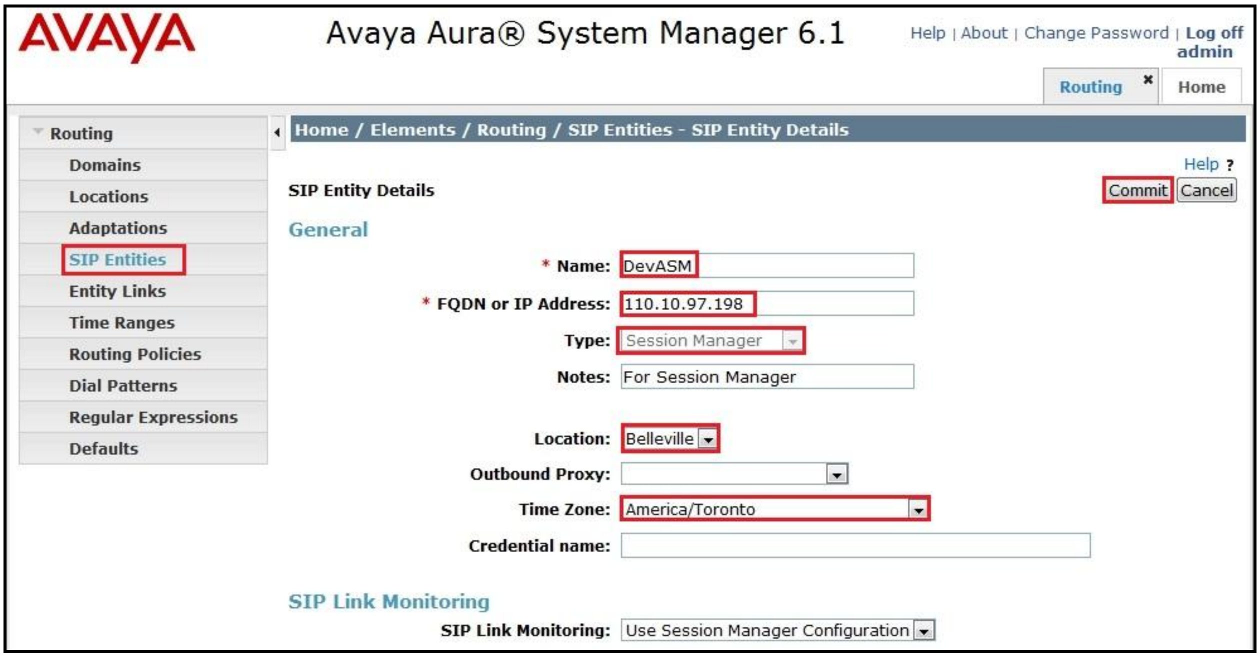
To add a new SIP Entity, navigate to **Routing → SIP Entities** in the left navigation pane and click on the **New** button in the right pane (not shown).

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

- **Name**: Enter a descriptive name

- **FQDN or IP Address**: Enter the FQDN or IP address of the SIP Entity that is used for SIP signaling
- **Type**: Select **Session Manager** for Session Manager and select **Other** for CS1000 and Acme SBC
- **Location**: Select one of the locations defined previously in **Section 6.3**
- **Time Zone**: Select the time zone for the location above
- Click **Commit** button.

The following screen shows the addition of a Session Manager SIP Entity. The IP address of the Session Manager signaling interface is entered for **FQDN or IP Address**.




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

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

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

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

The compliance test used **Port** 5060 with TCP for connecting to the CS1000 and the Acme SBC. It is shown in the screenshot below.

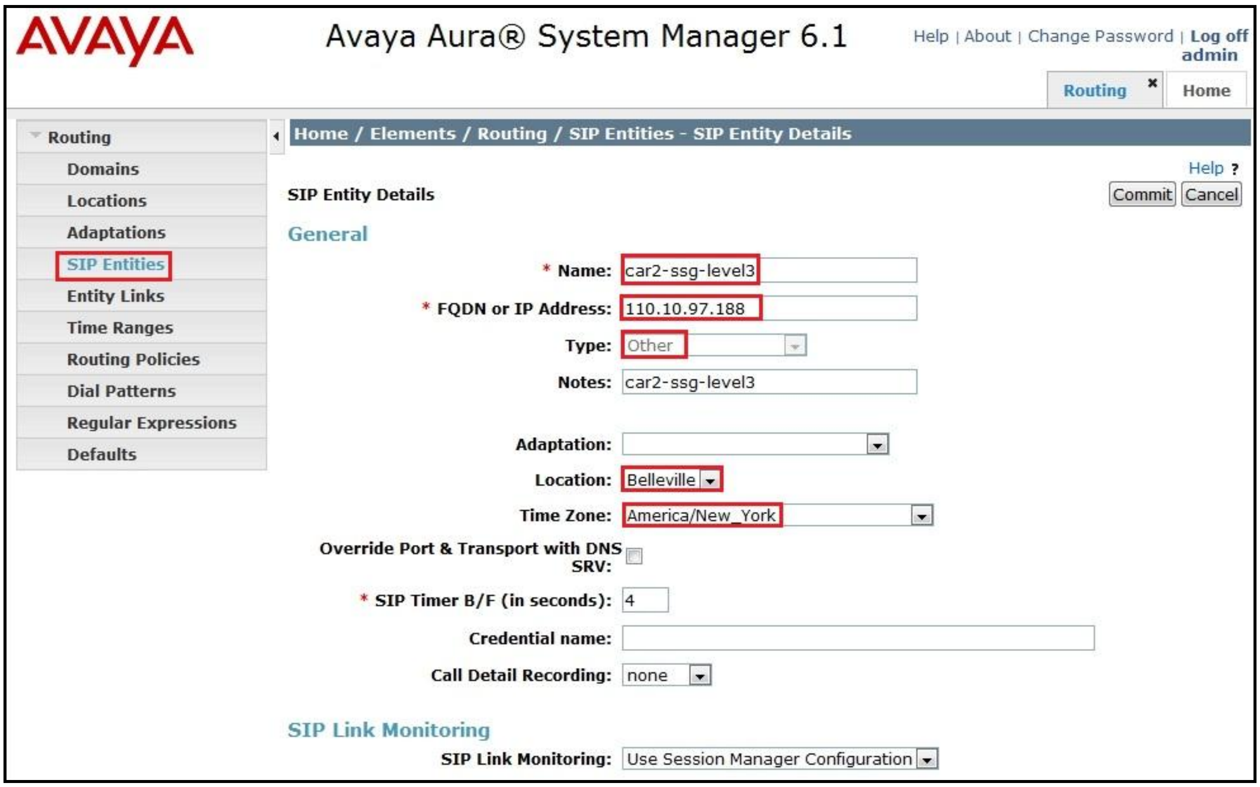
Port

Add Remove

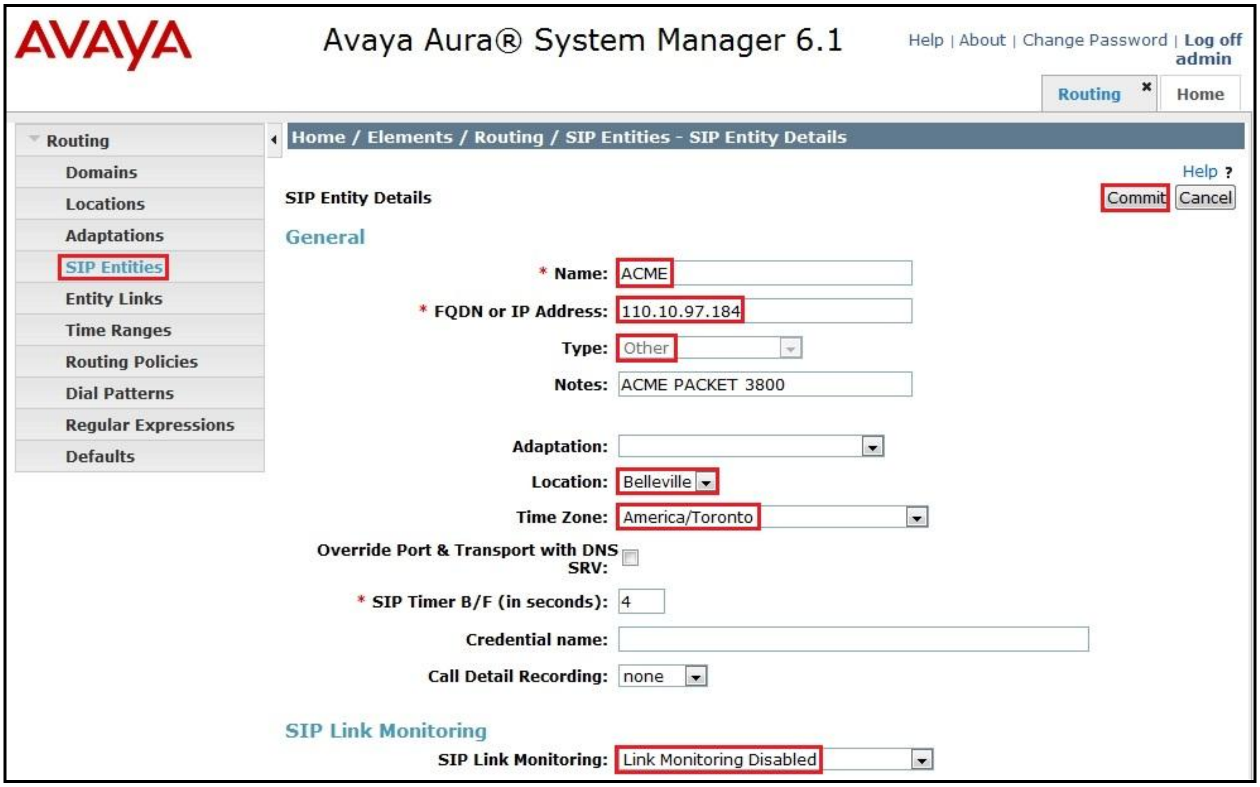
9 Items Refresh Filter: Enable

| | Port | Protocol | Default Domain | Notes |
|---|---|---|---|---|
| | 15060 | TLS | bvwdev.com | |
| | 5060 | UDP | bvwdev.com | |
| | 5060 | TCP | level3.com | |
| | 5061 | TLS | bvwdev.com | |

The following screen shows the addition of the CS1000 in the **SIP Entities** section. In order for Session Manager to send SIP traffic on an entity link to the CS1000, it is necessary to create a SIP Entity. The **FQDN or IP Address** field is set to the Node IP address of the CS1000 (see **Section 5.2.1**). Select **Type** as **Other**.




The following screen shows the addition of the SIP Entity for the Acme SBC. The **FQDN or IP Address** field is set to the IP address of its private network interface (see **Figure 1**). Select **Type** as **Other**. SIP Link Monitoring is disabled to prevent OPTIONS from being sent by Session Manager to the Acme SBC, Session Manager will use ICMP ping to monitor status of the SIP Trunk instead.




## 6.5. Add Entity Links

A SIP trunk between Session Manager and a telephony system is described by an Entity Link. Entity Links are created for the CS1000 and for the Acme SBC. To add an Entity Link, navigate to **Routing → Entity Links** in the left navigation pane and click the **New** button in the right pane (not shown).

Fill in the following fields in the new row that is displayed:

- **Name:** Enter a descriptive name
- **SIP Entity 1:** Select the Session Manager
- **Protocol:** Select the transport protocol used for this link
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end. For CS1000, this must match the port of **Proxy Server Route 1** which defined in **Section 5.5.2** step d)
- **SIP Entity 2:** Select the name of the other system. For CS1000 select the CS1000 SIP Entity; for the Acme SBC, select the Acme SBC SIP Entity. The SIP Entities are defined in **Section 6.4**
- **Port:** Port number on which the other system receives SIP requests from the Session Manager. For CS1000, this must match the **Local SIP Port** defined in **Section 5.5.2** step c)
- **Connection Policy:** Select **Trusted**. **Note**: If this is not selected, calls from the associated SIP Entity specified in **Section 6.4** will be denied
- Click **Commit** button to save

The following screens illustrate the Entity Links to the CS1000 and the Acme SBC. For the compliance test, transport protocol TCP and port 5060 are used to match the values of **Proxy Server Route 1** defined in **Section 5.5.2** step d) and in **Figure 1**.

Entity Link to CS1000:

**Entity Links**

Add Remove

1 Item | Refresh                                                   Filter: Enable

| | SIP Entity 1 | Protocol | Port | SIP Entity 2 | Port | Connection Policy |
|---|---|---|---|---|---|---|
| | DevASM | TCP | * 5060 | car2-ssg-level3 | * 5060 | Trusted |

Select : All, None

Entity Link to Acme SBC:

**Entity Links**

Add Remove

1 Item | Refresh                                                   Filter: Enable

| | SIP Entity 1 | Protocol | Port | SIP Entity 2 | Port | Connection Policy |
|---|---|---|---|---|---|---|
| | DevASM | UDP | * 5060 | ACME | * 5060 | Trusted |

Select : All, None

## 6.6. Add Routing Policies

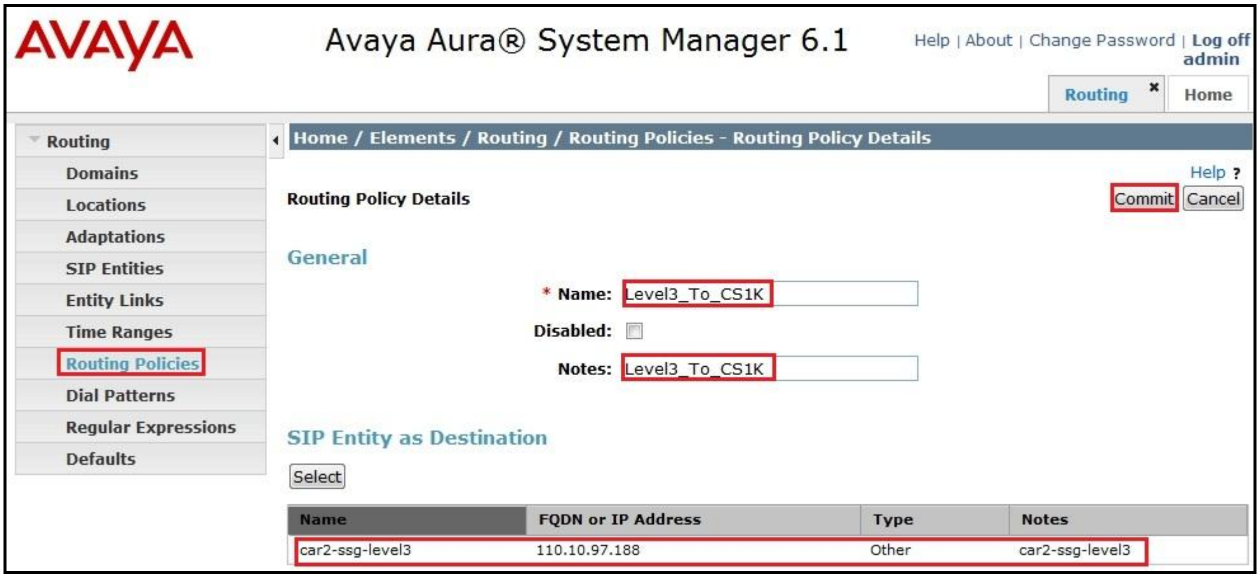
Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.5**. Routing policies must be added for the CS1000 and for the Acme SBC. To add a routing policy, navigate to **Routing → Routing Policies** in the left navigation pane and click **New** button in the right pane (not shown). The following screen is displayed. Fill in the following:

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

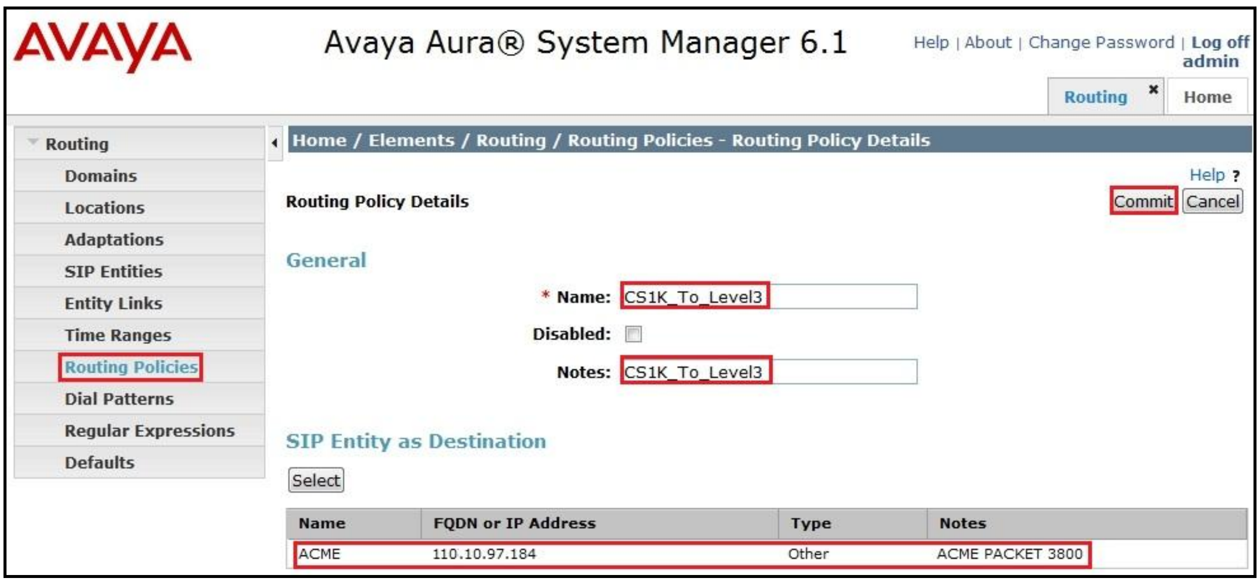
- **Name:** Enter a descriptive name
- **Notes:** Add a brief description (optional)

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

The following screens show the Routing Policies **Level3_To_CS1K** for the CS1000.




The following screens show the Routing Policies **CS1K_To_Level3** for the Acme SBC.




## 6.7. Add Dial Patterns

Dial Patterns are needed to route specific calls through Session Manager. For the compliance test, dial patterns are needed to route calls from CS1000 to Level 3 and vice versa. Dial Patterns define which route policy will be selected for a particular call based on the dialed digits, destination domain and originating location. To add a dial pattern, navigate to **Routing → Dial Patterns** in the left navigation pane and click on the **New** button in the right pane (not shown). Fill in the following, as shown in the screens below:

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

- **Pattern:** Enter a dial string that will be matched against the Request-URI of the call

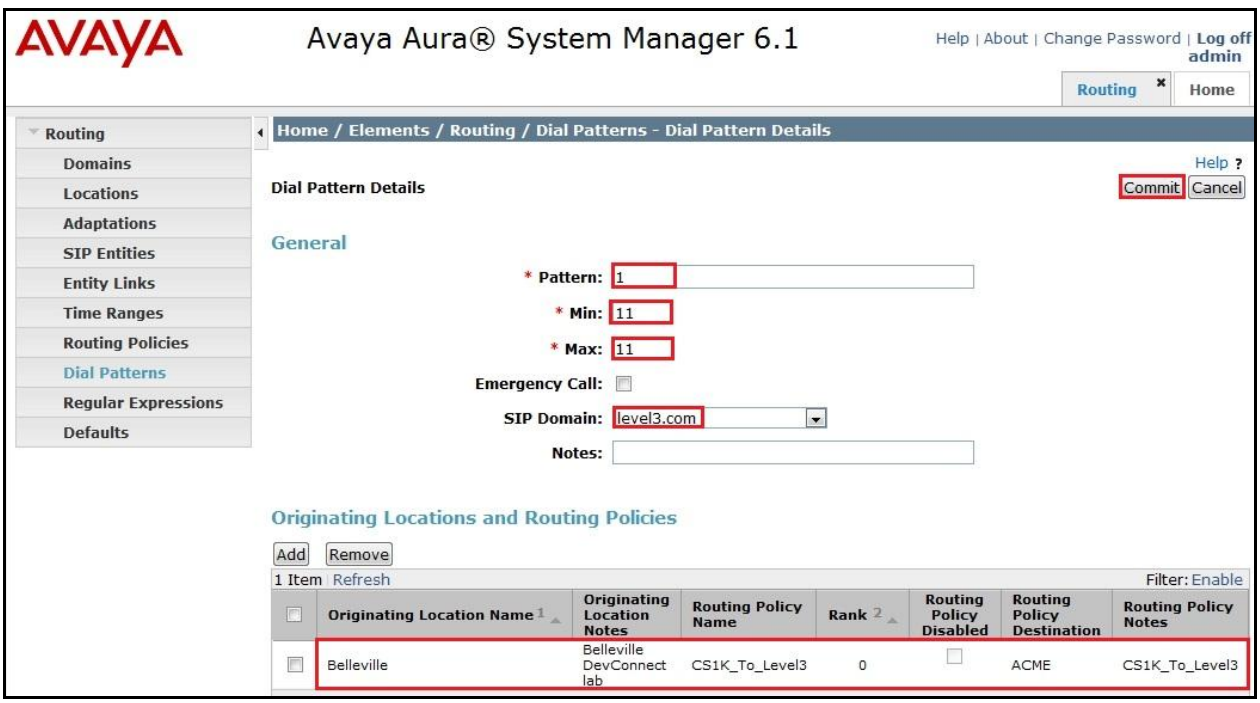
- **Min:** Enter a minimum length used in the match criteria
- **Max:** Enter a maximum length used in the match criteria
- **SIP Domain:** Enter the destination domain used in the match criteria
- **Notes:** Add a brief description (optional)

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

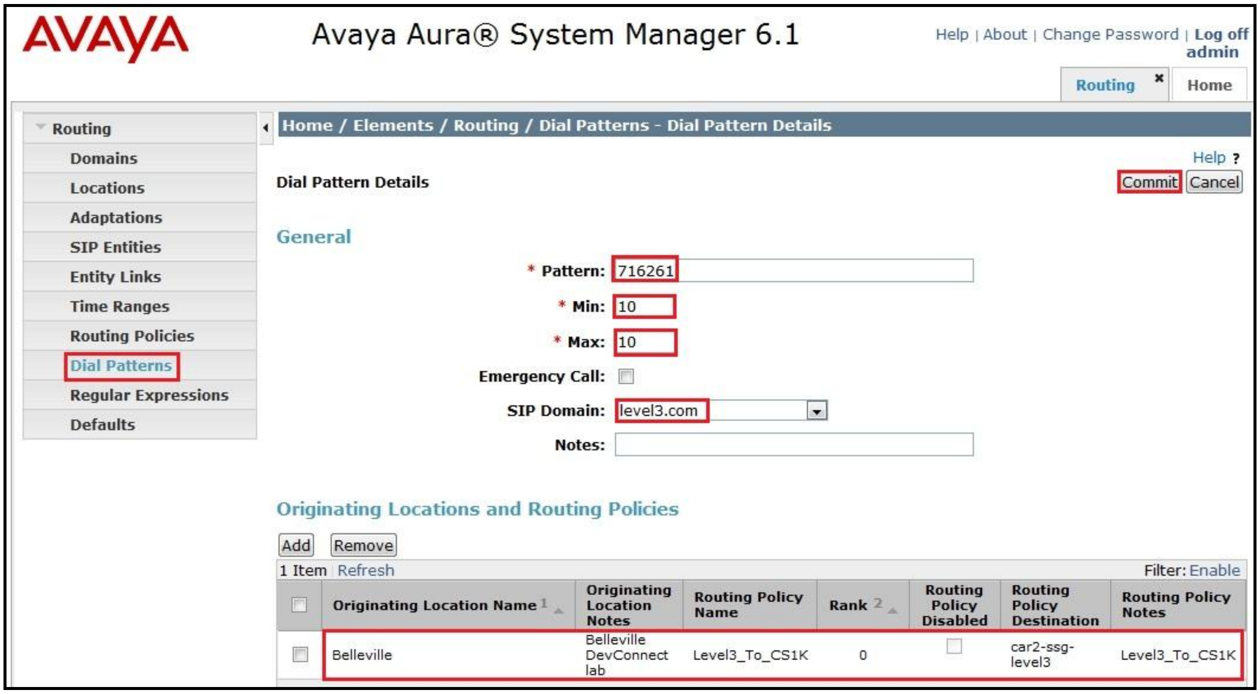
Default values can be used for the remaining fields. Click the **Commit** button to save.

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.

The first example in the screen below shows dial pattern for outbound 11-digit numbers. The dialed number starts with prefix 1 and has a destination domain of *level3.com* and uses route policy **CS1K_To_Level3** as defined in **Section 6.6**. The dial patterns for outbound calls that start with prefix 0 or 411 are configured similarly to this dial pattern.




The second example in the screen below shows dial pattern for inbound 10-digit numbers. The dialed number starts with prefix **716261** to domain **level3.com** and uses route policy **Level3_To_CS1000** as defined in **Section 6.6**. These are the DID numbers assigned to the enterprise by Level 3.



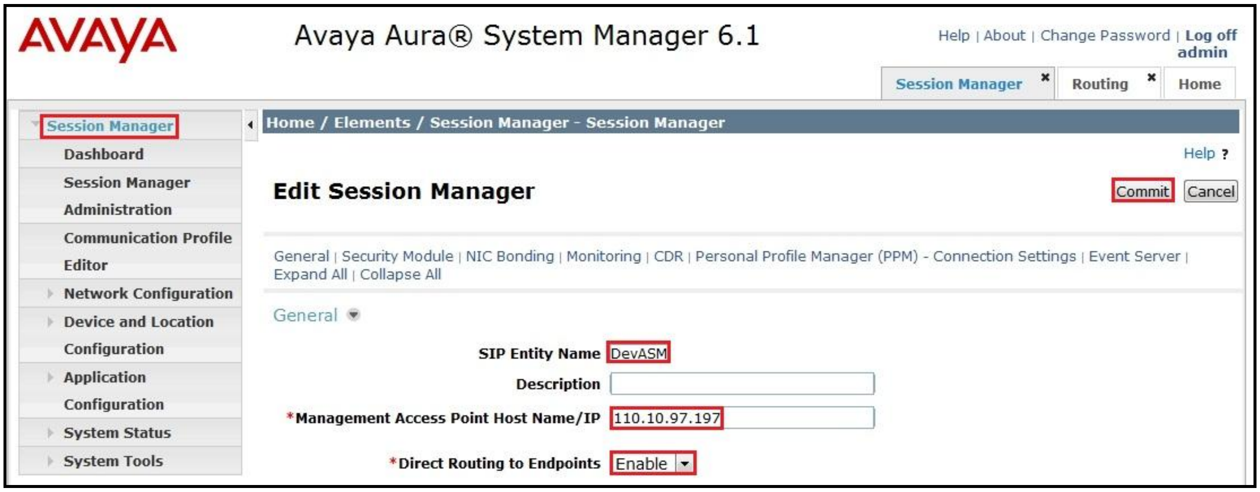

## 6.8. Add/View Session Manager

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

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

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

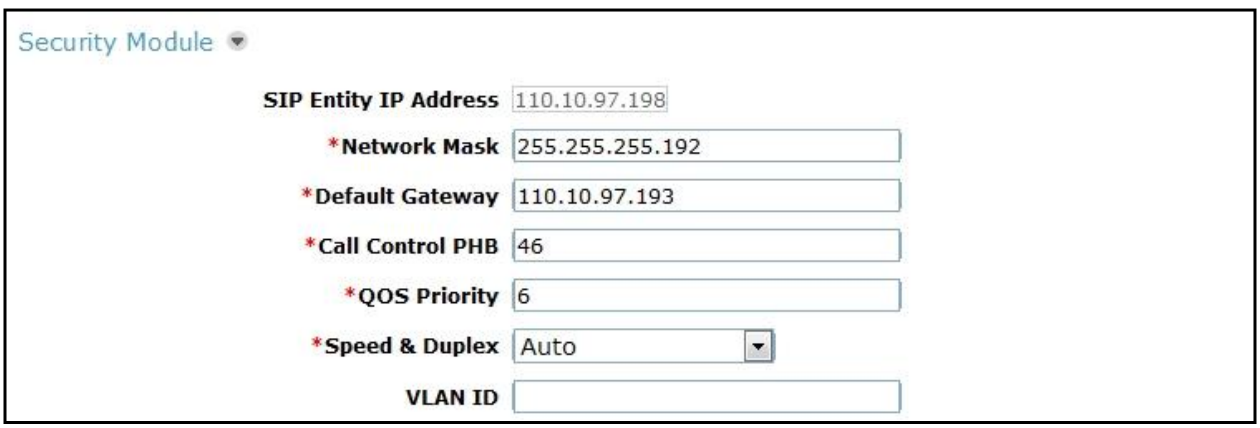
The screen below shows the Session Manager values used for the compliance test.




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

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

Use default values for the remaining fields. Click **Save** button (not shown) to add a Session Manager server. The screen below shows the remaining Session Manager values used for the compliance test.




# 7. Configure Acme Packet Net-Net 3800 Session Border Controller

This section describes the configuration of the Acme Packet Session Border Controller (SBC) necessary for interoperability with Avaya SIP-enabled enterprise solution and Level 3 SIP Trunking Service. The SBC is configured via the Acme Packet Command Line Interface (ACLI).

This section will not attempt to describe each component in its entirety, but instead will highlight fields in each component which relates to the functionality in these Application Notes. The remaining fields are generally the default/standard value pre-defined by the SBC.

In the compliance test, according to the recommended configuration in **Figure 1**, the enterprise network resides on the inside and the service provider resides on the outside of the SBC.

## 7.1. Acme Packet Command Line Interface

The SBC is configured using the ACLI. The following are the generic ACLI steps for configuring various elements.

1. Access to the console port of the SBC using a PC and a terminal emulation program such as HyperTerminal (use the RJ-45 to DB9 adapter as packaged with the SBC for cable connection).

   Use the following settings for the serial port on the PC.
   - Bits per second: 115200
   - Data bits: 8
   - Parity: None
   - Stop bits: 1
   - Flow control: None

2. Log into the SBC with the proper user password.
3. Enable the super user mode by entering **enable** command with a proper super user password. The command prompt will change to include a "#" instead of a ">" while in super user mode. This level of system access (i.e. at the "acmesystem#" prompt) will be referred to as the ***main*** level of the ACLI.
4. In super user mode, enter **configure terminal** command to access the system level where all operating and system elements may be configured. This level of system access will be referred to as the ***configuration*** level.
5. Enter the name of an element to be configured (e.g., **system**).
6. Enter the name of a sub-element, if any (e.g., **phy-interface).**
7. Enter the name of an element parameter followed by its value (e.g., **name INSIDE**).
8. Enter **done** to save changes to the element. Use of the **done** command causes the system to save and display the settings for the current element.
9. Enter **exit** as many times as necessary to return to the configuration level.
10. Repeat **Steps 5 - 9** to configure all other elements.
11. Enter **exit** to return to the main level.
12. Type **verify** to verify the configuration.
13. Type **save-config** to save the configuration.
14. Type **activate-config** to activate the configuration.

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

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

## 7.2. Physical and Network Interfaces

As part of the compliance test, the Ethernet interface of slot 1/port 0 of the SBC as shown below. It connects to the external public internet which is an un-trusted network.

```
phy-interface
        name                                   OUTSIDE
        operation-type                         Media
        port                                          0
        slot                                          1
        virtual-mac
        admin-state                            enabled
        auto-negotiation                       enabled
        duplex-mode                            FULL
        speed                                         100
        overload-protection                    disabled
```

The Ethernet interface slot 0/port 0 is connected to the internal corporate LAN as shown in the screen below:

```
phy-interface
        name                                   INSIDE
        operation-type                         Media
        port                                          0
        slot                                          0
        virtual-mac
        admin-state                            enabled
        auto-negotiation                       enabled
        duplex-mode                            FULL
        speed                                         100
        overload-protection                    disabled
```

Define a logical network interface for each physical interface to assign it a routable IP address. As described in **Figure 1**, the network interface below defines the IP addresses on the physical interface INSIDE which connects to the enterprise network.

```
network-interface
        name                                   INSIDE
        sub-port-id                            0
        description
        hostname
        ip-address                             110.10.97.184
        pri-utility-addr
        sec-utility-addr
        netmask                                255.255.255.192
        gateway                                110.10.97.129
        sec-gateway
        gw-heartbeat
                state                                       disabled
                heartbeat                                   0
                retry-count                                 0
                retry-timeout                               1
                health-score                                0
```

```
        dns-ip-primary
        dns-ip-backup1
        dns-ip-backup2
        dns-domain
        dns-timeout                           11
        hip-ip-list                           110.10.97.184
        ftp-address
        icmp-address                          110.10.97.184
        snmp-address
        telnet-address
```

The network interface below defines the IP addresses on physical interface OUTSIDE which connects to Level 3.

```
network-interface
        name                                  OUTSIDE
        sub-port-id                           0
        description
        hostname
        ip-address                            110.10.98.98
        pri-utility-addr
        sec-utility-addr
        netmask                               255.255.255.224
        gateway                               110.10.98.97
        sec-gateway
        gw-heartbeat
                state                                      disabled
                heartbeat                                  0
                retry-count                                0
                retry-timeout                              1
                health-score                               0
        dns-ip-primary
        dns-ip-backup1
        dns-ip-backup2
        dns-domain
        dns-timeout                           11
        hip-ip-list                           110.10.98.98
        ftp-address
        icmp-address                          110.10.98.98
        snmp-address
```

## 7.3. Realm

A realm represents a group of related SBC components.

For the compliance test, two realms are created. The realm name INSIDE represents the internal network which contains the elements configured for the enterprise.

```
realm-config
        identifier                            INSIDE
        description
        addr-prefix                           0.0.0.0
        network-interfaces
                                                     INSIDE:0

<Text removed for brevity>
```

The realm name OUTSIDE represents the external network which contains the elements configured for Level 3

```
realm-config
        identifier                             OUTSIDE
        description
        addr-prefix                            0.0.0.0
        network-interfaces
                                                      OUTSIDE:0
<Text removed for brevity>
```

## 7.4. Session Agent

A session agent defines the characteristics of signaling from a peer gateway endpoint such as Session Manager (as known as Call Server) or Level 3 (as known as Trunk Server).

The **session agent** in the screen below represents the configuration for Level 3. As described in **Figure 1**, the IP interface of Level 3 SIP Trunking Service is defined with transport protocol is UDP and port 5070.

- Set **state** to **enabled**
- Set **app-protocol** to **SIP**
- Set **realm-id** to **OUTSIDE**
- Set **in-manipulationid** to **Level3_To_CS1K**. This profile is defined in the SIP Header Manipulation Section as discussed later in **Section 7.7**. It is a set of rules to manipulate the SIP signaling for an inbound call from Level 3 such as to normalize the From, To, Request-URI headers etc. known to the CS1000.
- Set **out-manipulationid** to **CS1K_To_Level3**. This profile is defined in the SIP Header Manipulation Section as discussed later in **Section 7.7**. It is a set of rules to manipulate the SIP signaling for an outbound call to Level 3 such as to normalize the From, To, Request-URI headers etc. known to Level 3.

```
session-agent
        hostname                        220.20.35.91
        ip-address                       220.20.35.91
        port                                   5070
        state                                  enabled
        app-protocol                     SIP
        app-type
        transport-method                UDP
        realm-id                        OUTSIDE
        egress-realm-id
        description                     CS1K_To_Level3
             <Text removed for brevity>

        ping-method
        ping-interval
             <Text removed for brevity>

        in-manipulationid               Level3_To_CS1K
        out-manipulationid              CS1K_To_Level3
             <Text removed for brevity>
```

The **session agent** in the screen below represents the configuration for Session Manager. As described in **Figure 1**, the IP interface of Session Manager is defined with transport protocol is TCP and port 5060.

- Set **state** to **enabled**
- Set **app-protocol** to **SIP**
- Set **realm-id** to **INSIDE**

**Note**: the **in-manipulationid** and **out-manipulationid** are kept default which is blank. It means there is no signaling manipulation performed on the SIP traffic toward the CS1000. The manipulation is already applied to the Trunk Server side.

```
session-agent
        hostname                        110.10.97.198
        ip-address                      110.10.97.198
        port                            5060
        state                           enabled
        app-protocol                    SIP
        app-type
        transport-method                DynamicTCP
        realm-id                        INSIDE
        egress-realm-id
        description                     Level3_To_CS1K
             <Text removed for brevity>
```

## 7.5. SIP Configuration

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

Configure the sip-config as show in the screen below:

- Set the **state** to **enabled** to allow SIP call to be processed by the SBC
- Set **home-realm-id** to **INSIDE**
- Set **egress-realm-id** to **OUTSIDE**

```
sip-config
        state              enabled
        operation-mode            dialog
        dialog-transparency             enabled
        home-realm-id             INSIDE
        egress-realm-id           OUTSIDE
        nat-mode                  None
             <Text removed for brevity>
```

## 7.6. SIP Interface

SIP interface (***sip-interface***) enables the SIP application protocol on a particular network interface.

Two SIP interfaces are defined for this compliance test. The SIP interface as shown below is used by the SBC to listen to the enterprise SIP traffic from realm INSIDE. The SBC is configured to listen on network interface 110.10.97.184, transport protocol TCP and port 5060.

```
sip-interface
```

```
        state              enabled
        realm-id                   INSIDE
        description
        sip-port
                address           110.10.97.184
                port                     5060
                transport-protocol       TCP
       <Text removed for brevity>
```

The SIP interface shown below is used by the SBC to listen to SIP traffic from the realm OUTSIDE defined for Level 3. The SBC is configured to listen on network interface 110.10.98.98, transport protocol UDP and port 5060.

```
sip-interface
        state              enabled
        realm-id                   OUTSIDE
        description
        sip-port
                address           110.10.98.98
                port                     5060
                transport-protocol       UDP
       <Text removed for brevity>
```

## 7.7. SIP Manipulation

SIP Header Manipulation Rules (HMR) are used to modify the SIP messages (if necessary) for interoperability between the CS1000 and Level 3.

In the compliance test, Level 3 requires the SIP signaling from the enterprise to meet its specification. For that purpose, HMRs are created for Session Agent which are defined for Level 3 in **Section 7.4**.

The HMR **CS1K_To_Level3** is added as shown in the screen below to apply to SIP messages from the CS1000 toward Level 3. It contains rules to perform the following:

- Header rule **manipRURI** replaces the private enterprise SIP domain in the Request-URI header by $REMOTE_IP (.e.g. 220.20.35.91) which is the IP address assigned by Level 3.
- Header rule **manipTo** replaces the private enterprise SIP domain in the To header by $REMOTE_IP (.e.g. 220.20.35.91) which is the IP address assigned by Level 3.
- Header rule **manipFrom** replaces the private enterprise SIP domain in the From header by $LOCAL_IP (.e.g. 110.10.98.98) which is the public IP address of the Acme SBC.
- Header rule **manipPAI** replaces the private enterprise SIP domain in the P-Asserted-Identity (PAI) header by $LOCAL_IP (.e.g. 110.10.98.98) which is the public IP address of Acme SBC.
- Level 3 requires Diversion header for call forward scenarios as it does not support History-Info header. Therefore, a header rule **HistoryInfoRegex** was created to check if the History-Info header has a specific “reason code” to match the condition of off-net call forward scenarios. If a positive match happens, **AddDiversion1**, **AddDiversion2** and **AddDiversion3** will construct a Diversion header appropriate to call forward all call, busy and no answers

- Header rule **storePAI** stores calling party number  in the PAI header which will be used to construct a Remote-Party-ID header.
- Header rule **chkPrivacy** checks the **Privacy** header for a private call. This condition will be used by header rules **addRPID1**, **addRPID2** and **addRPID3** to construct the Remote-Party-ID from the PAI header value which is stored in the **storePAI** rule. This modification is to meet a requirement of Level 3 with respect to the SIP Trunk because Level 3 uses Remote-Party-ID header as an alternative to a PAI header.
- Header rule **delete_PAI** deletes the PAI header not required by Level 3.
- Header rule **delete_mcdn** deletes the mcdn body parts which are proprietary to the CS1000 and not required by Level 3.
- Header rule **delete_x_nt_e164_clid** deletes the X-nt-e164-clid header  which is proprietary to the CS1000 and not required by Level 3.
- Header rule **delete_Alert_Info** deletes the Alert_Info header  which is proprietary to the CS1000 and not required by Level 3.
- Header rule **delete_P_Location** deletes the P_Location header that is proprietary to Avaya and not required by Level 3.
- Header rule **delete_History_Info** deletes the History-Info header not required by Level 3.
- Header rule **delete_Route** deletes Route headers not required by Level 3.

```
sip-manipulation
        name                          CS1K_To_Level3
        description                   CS1K_To_Level3
        split-headers
        join-headers
        header-rule
                name                          manipRURI
                header-name                   request-uri
                action                        manipulate
                comparison-type               case-sensitive
                msg-type                      request
                methods                       INVITE,UPDATE
                match-value
                new-value
                element-rule
                        name                          modURIHost
                        parameter-name
                        type                          uri-host
                        action                        replace
                        match-val-type                any
                        comparison-type               case-sensitive
                        match-value
                        new-value                     $REMOTE_IP
       header-rule
                name                          manipTo
                header-name                   To
                action                        manipulate
                comparison-type               case-sensitive
                msg-type                      request
                methods                       INVITE
                match-value
                new-value
                element-rule
                        name                          modTo
                        parameter-name
```

```
                type                            uri-host
                action                          replace
                match-val-type                  any
                comparison-type                 case-sensitive
                match-value
                new-value                       $REMOTE_IP
header-rule
        name                            manipFrom
        header-name                     From
        action                          manipulate
        comparison-type                 case-sensitive
        msg-type                        request
        methods                         INVITE
        match-value
        new-value
        element-rule
                name                            modFrom
                parameter-name
                type                            uri-host
                action                          replace
                match-val-type                  any
                comparison-type                 case-sensitive
                match-value
                new-value                       $LOCAL_IP

header-rule
        name                            manipPAI
        header-name                     P-Asserted-Identity
        action                          manipulate
        comparison-type                 case-sensitive
        msg-type                        any
        methods                         INVITE
        match-value
        new-value
        element-rule
                name                            modPAI
                parameter-name
                type                            uri-host
                action                          replace
                match-val-type                  any
                comparison-type                 case-sensitive
                match-value
                new-value                       $LOCAL_IP

header-rule
        name                            HistoryInfoRegex
        header-name                     History-Info
        action                          store
        comparison-type                 pattern-rule
        msg-type                        any
        methods
        match-value                     ()
        new-value
        element-rule
                name                            GetUser
                parameter-name
                type                            uri-user
                action                          store
                match-val-type                  any
                comparison-type                 pattern-rule
                match-value
                new-value
```

```
                element-rule
                        name                            GetHost
                        parameter-name
                        type                            uri-host
                        action                          store
                        match-val-type                  any
                        comparison-type                 pattern-rule
                        match-value
                        new-value
                element-rule
                        name                            GetUserReason1
                        parameter-name
                        type                            header-value
                        action                          store
                        match-val-type                  any
                        comparison-type                 pattern-rule
                        match-value                     (.*)(Moved)(.*)
                        new-value
                element-rule
                        name                            GetUserReason2
                        parameter-name
                        type                            header-value
                        action                          store
                        match-val-type                  any
                        comparison-type                 pattern-rule
                        match-value                     (.*)(Busy)(.*)
                        new-value
                element-rule
                        name                            GetUserReason3
                        parameter-name
                        type                            header-value
                        action                          store
                        match-val-type                  any
                        comparison-type                 pattern-rule
                        match-value                     (.*)(Unavailable)(.*)
                        new-value
        header-rule
                name                            AddDiversion1
                header-name                     Diversion
                action                          add
                comparison-type                 boolean
                msg-type                        any
                methods
                match-value                     $HistoryInfoRegex[0].$GetUserReason1
                new-value
<sip:+$HistoryInfoRegex[0].$GetUser.$0+@+$HistoryInfoRegex[0].$GetHost.$0+>;reason=unc
onditional;screen=no
        header-rule
                name                            AddDiverion2
                header-name                     Diversion
                action                          add
                comparison-type                 boolean
                msg-type                        any
                methods
                match-value                     $HistoryInfoRegex[0].$GetUserReason2
                new-value
<sip:+$HistoryInfoRegex[0].$GetUser.$0+@+$HistoryInfoRegex[0].$GetHost.$0+>;reason=use
r\-busy;screen=no
        header-rule
                name                            AddDiversion3
                header-name                     Diversion
                action                          add
```

```
                comparison-type                 boolean
                msg-type                        any
                methods
                match-value                     $HistoryInfoRegex[0].$GetUserReason3
                new-value
<sip:+$HistoryInfoRegex[0].$GetUser.$0+@+$HistoryInfoRegex[0].$GetHost.$0+>;reason=no\
-answer;screen=no
        header-rule
                name                            chkPrivacy
                header-name                     Privacy
                action                          store
                comparison-type                 pattern-rule
                msg-type                        any
                methods
                match-value
                new-value
                element-rule
                        name                            privacyNone
                        parameter-name
                        type                            header-value
                        action                          store
                        match-val-type                  any
                        comparison-type                 pattern-rule
                        match-value                     ^none$
                        new-value
                element-rule
                        name                            privacyID
                        parameter-name
                        type                            header-value
                        action                          store
                        match-val-type                  any
                        comparison-type                 pattern-rule
                        match-value                     ^id$
                        new-value
                element-rule
                        name                            privacyUser
                        parameter-name
                        type                            header-value
                        action                          store
                        match-val-type                  any
                        comparison-type                 pattern-rule
                        match-value                     ^user$
                        new-value
                element-rule
                        name                            privacyIDUser
                        parameter-name
                        type                            header-value
                        action                          store
                        match-val-type                  any
                        comparison-type                 pattern-rule
                        match-value                     ^id;user$
                        new-value

        header-rule
                name                            storePAI
                header-name                     P-Asserted-Identity
                action                          store
                comparison-type                 case-sensitive
                msg-type                        any
                methods
                match-value
                new-value
```

```
                element-rule
                        name                              storeHeader
                        parameter-name
                        type                              header-value
                        action                            store
                        match-val-type                    any
                        comparison-type                   case-sensitive
                        match-value
                        new-value
        header-rule
                name                              addRPID1
                header-name                       Remote-Party-ID
                action                            add
                comparison-type                   boolean
                msg-type                          any
                methods
                match-value                       $checkPrivacy[0].$privacyNone
                new-value
$storePAI[0].$storeHeader.$0+";screen=no;privacy=off"
        header-rule
                name                              addRPID2
                header-name                       Remote-Party-ID
                action                            add
                comparison-type                   boolean
                msg-type                          any
                methods
                match-value                       $checkPrivacy[0].$privacyID
                new-value
$storePAI[0].$storeHeader.$0+";screen=no;privacy=on"
        header-rule
                name                              addRPID3
                header-name                       Remote-Party-ID
                action                            add
                comparison-type                   boolean
                msg-type                          any
                methods
                match-value                       $checkPrivacy[0].$privacyUser
                new-value
$storePAI[0].$storeHeader.$0+";screen=no;privacy=on"
        header-rule
                name                              addRPID4
                header-name                       Remote-Party-ID
                action                            add
                comparison-type                   case-sensitive
                msg-type                          any
                methods
                match-value                       $checkPrivacy[0].$privacyIDUser
                new-value
$storePAI[0].$storeHeader.$0+";screen=no;privacy=on"
        header-rule
                name                              delete_PAI
                header-name                       P-Asserted-Identity
                action                            delete
                comparison-type                   case-sensitive
                msg-type                          any
                methods
                match-value
                new-value
        header-rule
                name                              delete_mcdn
                header-name                       Content-Type
                action                            manipulate
```

```
          comparison-type                 case-sensitive
          msg-type                        any
          methods
          match-value
          new-value
          element-rule
                  name                            delete_nt_epid
                  parameter-name                  application/x-nt-epid-frag-hex
                  type                            mime
                  action                          delete-element
                  match-val-type                  any
                  comparison-type                 case-sensitive
                  match-value
                  new-value
          element-rule
                  name                            delete_nt_mcdn
                  parameter-name                  application/x-nt-mcdn-frag-hex
                  type                            mime
                  action                          delete-element
                  match-val-type                  any
                  comparison-type                 case-sensitive
                  match-value
                  new-value
  header-rule
          name                            delete_x_nt_e164_clid
          header-name                     X-nt-e164-clid
          action                          delete
          comparison-type                 case-sensitive
          msg-type                        any
          methods
          match-value
          new-value
  header-rule
          name                            delete_Alert_Info
          header-name                     Alert_info
          action                          delete
          comparison-type                 case-sensitive
          msg-type                        any
          methods
          match-value
          new-value
  header-rule
          name                            delete_P_Location
          header-name                     P-Location
          action                          delete
          comparison-type                 case-sensitive
          msg-type                        any
          methods
          match-value
          new-value
  header-rule
          name                            delete_History_Info
          header-name                     History-Info
          action                          delete
          comparison-type                 case-sensitive
          msg-type                        any
          methods
          match-value
          new-value
  header-rule
          name                            del_Route
          header-name                     Route
```

```
                action                          delete
                comparison-type                 case-sensitive
                msg-type                        any
                methods
                match-value
                new-value
```

The HMR **Level3_To_CS1K** in the screen below is created for the Session Agent which is defined for Level 3 in **Section 7.4**. The HMR is applied to SIP messages from Level 3 toward the CS1000. It contains rules to perform the following:

- Header rule **manipRURI** replaces the IP address in the Request-URI header with the domain name **level3.com** expected by the CS1000.
- Header rule **manipTo** replaces the IP address in the To header with the domain name **level3.com** expected by the CS1000.
- Header rule **manipFrom** replaces the IP address in the From header with the domain name **level3.com** expected by the CS1000.
- Header rule **manipAllow** removes UPDATE from the Allow header. This prevents the CS1000 from using UPDATE on the SIP Trunk. This implementation is to support blind transfer off-net scenario when Level 3 does not fully support the UPDATE method. For detail information, please refer to **Section 2.2**, observation #02.

```
sip-manipulation
        name                            Level3_To_CS1K
        description                     Level3_To_CS1K
        split-headers
        join-headers
        header-rule
                name                            manipRURI
                header-name                     request-uri
                action                          manipulate
                comparison-type                 case-sensitive
                msg-type                        any
                methods
                match-value
                new-value
                element-rule
                        name                            modRURI
                        parameter-name
                        type                            uri-host
                        action                          replace
                        match-val-type                  any
                        comparison-type                 case-sensitive
                        match-value
                        new-value                       level3.com
        header-rule
                name                            manipTo
                header-name                     To
                action                          manipulate
                comparison-type                 case-sensitive
                msg-type                        any
                methods
                match-value
                new-value
                element-rule
                        name                            To
                        parameter-name
```

```
                type                              uri-host
                action                            replace
                match-val-type                    any
                comparison-type                   case-sensitive
                match-value
                new-value                         level3.com
        header-rule
                name                              manipFrom
                header-name                       From
                action                            manipulate
                comparison-type                   case-sensitive
                msg-type                          any
                methods
                match-value
                new-value
                element-rule
                        name                              From
                        parameter-name
                        type                              uri-host
                        action                            replace
                        match-val-type                    any
                        comparison-type                   case-sensitive
                        match-value
                        new-value                         level3.com
        header-rule
                name                              manipAllow
                header-name                       Allow
                action                            manipulate
                comparison-type                   case-sensitive
                msg-type                          any
                methods
                match-value
                new-value                         $ORIGINAL-",UPDATE"
```

## 7.8. Steering Pools

Steering pools define the range of ports to be used for RTP.

For the compliance test, separate steering pools are defined for each realm.

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

- **ip-address:** The network interface will be used to transmit or receive the RTP.
- **start-port:** An number that begins the port range for RTP.
- **end-port:** An number that ends the port range for RTP.
- **realm-id:** The realm to which steering pool is assigne.

The screen below is the steering pool for **OUTSIDE** realm:

```
steering-pool
        ip-address                110.10.98.98
        start-port                20000
        end-port                  40000
        realm-id                  OUTSIDE
<Text removed for brevity>
```

The screen below is the steering pool for **INSIDE** realm:

```
steering-pool
        ip-address                 10.10.97.184
        start-port                 20000
        end-port                   40000
        realm-id                   INSIDE
<Text removed for brevity>
```

## 7.9. Local Policy

The local policies govern the routing of a call from the enterprise to the service provider and vice versa.

Two local policies are created for the compliance test.

For inbound calls, the local-policy allows all calls from source realm OUTSIDE to pass through the Acme SBC.

To activate the local-policy, set the **state** to **enabled**

The policy-attribute is defined as follows:

- Set **from-address** to * (an asterisk character)
- Set **to-address** to * (an asterisk character)
- Set the **next-hop** to the IP address of Session Manager
- Set the **realm** to **INSIDE**
- Set the **app-protocol** to **SIP**
- Set the **state** to **enabled**

```
local-policy
        from-address
                                        *
        to-address
                                        *
        source-realm
                                        OUTSIDE
        description                     Level3_To_CS1K
        activate-time                   N/A
        deactivate-time                 N/A
        state                           enabled
        policy-priority                 none
       <Text removed for brevity>
        policy-attribute
                next-hop                        110.10.97.198
                realm                           INSIDE
                action                          none
                terminate-recursion             disabled
                carrier
                start-time                      0000
                end-time                        2400
                days-of-week                    U-S
                cost                            0
                app-protocol                    SIP
                state                           enabled
```

```
            methods
       <Text removed for brevity>
```

For outbound calls, the local-policy allows all calls from source realm **INSIDE** to any PSTN destination to pass through the Acme SBC.

To activate the local-policy, set the **state** to **enabled**

The policy-attribute is defined as follows:

- Set **from-address** to * (an asterisk character)
- Set **to-address** to * (an asterisk character)
- Set the **next-hop** to the IP address of the Level 3 SIP Trunking Service
- Set the **realm** to **OUTSIDE**
- Set the **app-protocol** to **SIP**
- Set the **state** to **enabled**

```
local-policy
        from-address
                                       *
        to-address
                                       *
        source-realm
                                       INSIDE
        description
        activate-time                  N/A
        deactivate-time                N/A
        state                          enabled
        policy-priority                none
       <Text removed for brevity>
        policy-attribute
                next-hop                       220.10.35.91
                realm                          OUTSIDE
                action                         none
                terminate-recursion            disabled
                carrier
                start-time                     0000
                end-time                       2400
                days-of-week                   U-S
                cost                           0
                app-protocol                   SIP
                state                          enabled
                methods
       <Text removed for brevity>
```

# 8. Level 3 SIP Trunking Service Configuration

Level 3 is responsible for the configuration of its SIP Trunking Service. The customer will need to provide the IP address used to reach the Acme SBC at enterprise side. Level 3 will provide the customer with the necessary information to configure the SIP connection from the enterprise to Level 3. The information provided by Level 3 includes:

- IP address of the Level 3 Session Border Controller

- Level 3 SIP domain. In the compliance test, Level 3 preferred to use an IP address as a URI-Host
- Enterprise SIP domain. In the compliance test, Level 3 preferred to use the IP address of the Acme SBC as a URI-Host
- Supported codecs
- DID numbers
- IP addresses and port numbers used for signaling or media through any security devices
- Digest Authentication information

The sample configuration between Level 3 and the enterprise for the compliance test is a static configuration. There is no registration on the SIP trunk implemented on either Level 3 or enterprise side.

# 9. Verification and Troubleshooting

This section provides verification steps that may be performed in the field to verify that the solution is configured properly. This section also provides a list of useful troubleshooting commands.

## 9.1. Verification Steps

The following activities are made to each test scenario.

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

## 9.2. Protocol Traces:

The following SIP headers are inspected using Wireshark traces:
- Request-URI: verify the request number and SIP domain
- From: verify the display name and display number

- To: verify the display name and display number
- Remote-Party-ID: verify the display name and display number
- Privacy: verify privacy masking with "user, id"
- Diversion: verify DID number
-Authorization: verify Digest Authentication

The following attributes in SIP message body are inspected using Wireshark traces:
- Connection Information (c line): verify IP address of near end and far end endpoints
- Time Description (t line): verify session timeout value of near end and far end endpoints
- Media Description (m line): verify audio port, codec, DTMF event description
- Media Attribute (a line): verify specific audio port, codec, ptime, send/ receive ability, DTMF event and fax attributes

## 9.3. Troubleshooting:

### 9.3.1.1 Acme SBC

Using a network sniffing tool (e.g., Wireshark) to monitor the SIP signaling messages between Level 3 and Acme SBC

The following is an example inbound call from Level 3 to the CS1000.
- Inbound INVITE request from Level 3:

```
INVITE sip:7162611205@110.10.98.98:5060 SIP/2.0
Via: SIP/2.0/UDP 220.20.35.91:5070;branch=z9hG4bKdhms3h30a0o04kkv46c1.1
From: <sip:6139675258@220.20.35.91;user=phone>;tag=SDcsife01-334911229-
1317654395748-
To: "AVAYA ."<sip:7162611205@110.10.98.98>
Call-ID: SDcsife01-31188195e95bd3226c681c5c96ba95e5-v3000i1
CSeq: 623459763 INVITE
Contact: <sip:6139675258@220.20.35.91:5070;transport=udp>
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Accept: multipart/mixed,application/media_control+xml,application/sdp
Supported:
Max-Forwards: 9
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 281

v=0
o=BroadWorks 4903705 1 IN IP4 220.20.35.91
s=-
c=IN IP4 220.20.35.91
t=0 0
m=audio 49264 RTP/AVP 0 8 18 101
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=maxptime:20
```

- 200OK/SDP response by the CS1000:

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 220.20.35.91:5070;branch=z9hG4bKdhms3h30a0o04kkv46c1.1
From: <sip:6139675258@220.20.35.91;user=phone>;tag=SDcsife01-334911229-
1317654395748-
To: "AVAYA ."<sip:7162611205@110.10.98.98>;tag=633f438-bc610a87-13c4-55013-
3f78d-45e32a84-3f78d
Call-ID: SDcsife01-31188195e95bd3226c681c5c96ba95e5-v3000i1
CSeq: 623459763 INVITE
Supported: 100rel,x-nortel-sipvc,replaces
User-Agent: Nortel CS1000 SIP GW release_7.0 version_ssLinux-7.50.17
Privacy: none
Contact: <sip:7162611205@110.10.98.98:5060;user=phone;transport=udp>
Allow:
INVITE,ACK,BYE,REGISTER,REFER,NOTIFY,CANCEL,PRACK,OPTIONS,INFO,SUBSCRIBE,UPDATE
Content-Type: application/sdp
Content-Length: 271
Server: AVAYA-SM-6.1.1.0.611023
Remote-Party-ID: "Level3 i1120"
<sip:7162611205@220.20.35.91;user=phone>;screen=no;privacy=off

v=0
o=- 26 1 IN IP4 110.10.98.98
s=-
c=IN IP4 110.10.98.98
t=0 0
m=audio 20346 RTP/AVP 18 101 111
c=IN IP4 110.10.98.98
a=maxptime:20
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtpmap:111 X-nt-inforeq/8000
a=ptime:20
a=sendrecv
```

The following is an example outbound call from the CS1000 to Level 3.

- Outbound INVITE request from CS1000:

```
INVITE sip:16139675279@220.20.35.91;user=phone SIP/2.0
Via: SIP/2.0/UDP 110.10.98.98:5060;branch=z9hG4bKjed9ef30dgh0ng0sv531.1
From: "Level3 i1120" <sip:7162611205@110.10.98.98;user=phone>;tag=633dbb8-
bc610a87-13c4-55013-3f58c-111153c4-3f58c
To: <sip:16139675279@220.20.35.91;user=phone>
Call-ID: 7d6d638-bc610a87-13c4-55013-3f58c-38af4fc6-3f58c
CSeq: 1 INVITE
Supported: 100rel,x-nortel-sipvc,replaces
User-Agent: Nortel CS1000 SIP GW release_7.0 version_ssLinux-7.50.17 AVAYA-SM-
6.1.1.0.611023
Privacy: none
Contact: <sip:7162611205@110.10.98.98:5060;user=phone;transport=udp>
Allow:
INVITE,ACK,BYE,REGISTER,REFER,NOTIFY,CANCEL,PRACK,OPTIONS,INFO,SUBSCRIBE,UPDATE
Content-Type: application/sdp
Content-Length: 260
```

```
Max-Forwards: 65
Remote-Party-ID: "Level3 i1120"
<sip:7162611205@220.20.35.91;user=phone>;screen=no;privacy=off

v=0
o=- 25 1 IN IP4 110.10.98.98
s=-
c=IN IP4 110.10.98.98
t=0 0
m=audio 20344 RTP/AVP 0 8 18 101 111
c=IN IP4 110.10.98.98
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtpmap:111 X-nt-inforeq/8000
a=ptime:20
a=sendrecv
```

- 401 challenge from Level 3 to request Digest Authentication:

```
SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 110.10.98.98:5060;branch=z9hG4bKjed9ef30dgh0ng0sv531.1
From: "Level3 i1120" <sip:7162611205@110.10.98.98;user=phone>;tag=633dbb8-
bc610a87-13c4-55013-3f58c-111153c4-3f58c
To: <sip:16139675279@220.20.35.91;user=phone>;tag=SDp9aae99-1806987150-
1317653882939
Call-ID: 7d6d638-bc610a87-13c4-55013-3f58c-38af4fc6-3f58c
CSeq: 1 INVITE
WWW-Authenticate: DIGEST
qop="auth",nonce="BroadWorksXgtbl3n0bT4egtd9BW",algorithm=MD5,realm="BroadWorks"
Content-Length:
```

- Re-INVITE from the CS1000 with Authorization header responds to Digest Authentication:

```
INVITE sip:16139675279@220.20.35.91;user=phone SIP/2.0
Via: SIP/2.0/UDP 110.10.98.98:5060;branch=z9hG4bKk6ka0o30eg4hng8th160.1
From: "Level3 i1120" <sip:7162611205@110.10.98.98;user=phone>;tag=633dbb8-
bc610a87-13c4-55013-3f58c-111153c4-3f58c
To: <sip:16139675279@220.20.35.91;user=phone>
Call-ID: 7d6d638-bc610a87-13c4-55013-3f58c-38af4fc6-3f58c
CSeq: 2 INVITE
Supported: 100rel,x-nortel-sipvc,replaces
User-Agent: Nortel CS1000 SIP GW release_7.0 version_ssLinux-7.50.17 AVAYA-SM-
6.1.1.0.611023
Privacy: none
Contact: <sip:7162611205@110.10.98.98:5060;user=phone;transport=udp>
Authorization: Digest username="1-23Q-
3413",realm="BroadWorks",nonce="BroadWorksXgtbl3n0bT4egtd9BW",uri="sip:16139675
279@level3.com;user=phone",response="d78a793bf7ec494cf17bcc6e21e3e366",algorith
m=MD5,cnonce="f772dc7",qop=auth,nc=00000001
Allow:
INVITE,ACK,BYE,REGISTER,REFER,NOTIFY,CANCEL,PRACK,OPTIONS,INFO,SUBSCRIBE,UPDATE
Content-Type: application/sdp
Content-Length: 260
Max-Forwards: 65
```

```
Remote-Party-ID: "Level3 i1120"
<sip:7162611205@220.20.35.91;user=phone>;screen=no;privacy=off

v=0
o=- 25 1 IN IP4 110.10.98.98
s=-
c=IN IP4 110.10.98.98
t=0 0
m=audio 20344 RTP/AVP 0 8 18 101 111
c=IN IP4 110.10.98.98
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtpmap:111 X-nt-inforeq/8000
a=ptime:20
a=sendrecv
```

- 200OK/SDP response by Level 3:

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 110.10.98.98:5060;branch=z9hG4bKk6ka0o30eg4hng8th160.1
From: "Level3 i1120" <sip:7162611205@110.10.98.98;user=phone>;tag=633dbb8-
bc610a87-13c4-55013-3f58c-111153c4-3f58c
To: <sip:16139675279@220.20.35.91;user=phone>;tag=SDp9aae99-503571485-
1317653884615
Call-ID: 7d6d638-bc610a87-13c4-55013-3f58c-38af4fc6-3f58c
CSeq: 2 INVITE
Supported:
Contact: <sip:16139675279@220.20.35.91:5070;transport=udp>
Remote-Party-ID:
<sip:16139675279@8.13.220.254;user=phone>;screen=yes;party=called;privacy=off;id-
type=subscriber
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Accept: multipart/mixed,application/media_control+xml,application/sdp
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 210

v=0
o=BroadWorks 4902809 1 IN IP4 220.20.35.91
s=-
c=IN IP4 220.20.35.91
t=0 0
m=audio 49262 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=maxptime:20
```

#### 9.3.1.2 CS1000 Verification Steps

a) Verify patch installation on CS1000

Following screen shows the output of “dstat” command on Call Server:

```
pdt> dstat
Call Server:
------------
DepList name: core
        Filename: /var/opt/nortel/cs/fs/u/patch/deplist/mcore_01.cpl
        Issue   :  01
        Release :  x2107.50
        Created :  2012-05-16 12:51:18 (est)
        Number of patches: 215
        Patches Loaded: 215
        Patches In-service: 215
pdt>
```

Following screen shows the output of “spstat” command on SSG Server:

```
[admin@car2-sps ~]$ spstat
There is no SP in loaded status.
The last applied SP: Service_Pack_Linux_7.50_17_20120516.ntl
It is a STANDARD SP.
Has been applied by user nortel on Wed Jun  6 20:02:15 2012.
spins command completed with no errors detected.
[admin@car2-sps ~]$
```

b) Active Call Trace (LD 80)
The following is an example of one of the commands available on the CS1000 to trace the DN when the call is in progress.  The call scenario involved the PSTN phone number 6139675258 calling 7162611205 on CS1000.

- Login to the Call Server CLI (please refer to **Section 5.1.2** for more detail)
- Login to the Overlay command prompt, issue the command **LD 80** and then **trace 5 1205**
- After the call is released, issue the command **trac 5 1205** again to see if the DN is released back to idle state

Below is the actual output of the Call Server Command Line mode when the 1205 is in-call state:

```
>ld 80
TRA000
.trac 5 1205

ACTIVE  VTN 108 0 00 25

ORIG   VTN 104 1 00 00   VTRK IPTI  RMBR  105 1 INCOMING VOIP GW CALL
  FAR-END SIP SIGNALLING IP: 110.10.97.184
  FAR-END MEDIA ENDPOINT IP: 110.10.97.184  PORT: 21004
  FAR-END VendorID: AVAYA-SM-6.1.6.0.616008
TERM   VTN 108 0 00 20  KEY 0  SCR MARP  CUST 5  DN 1205  TYPE 1120
  SIGNALLING ENCRYPTION: INSEC
  MEDIA ENDPOINT IP: 110.10.98.141  PORT: 5200
MEDIA PROFILE: CODEC G.711 MU-LAW  PAYLOAD 20 ms  VAD OFF
RFC2833:  RXPT  101   TXPT  101   DIAL DN 1205
MAIN_PM  ESTD
TALKSLOT  ORIG  76   TERM  49
EES_DATA:
NONE
QUEU  NONE
CALL ID 0 34209
```

```
----  ISDN ISL CALL (ORIG) ----
CALL REF # =  385
BEARER CAP =  VOICE
HLC =
CALL STATE =  10     ACTIVE
CALLING NO =  6139675258  NUM_PLAN:UNKNOWN     TON:UNKNOWN   ESN:UNKNOWN
CALLED NO  =  7162611205  NUM_PLAN:UNKNOWN     TON:UNKNOWN   ESN:UNKNOWN
```

The following is an example after the call on 1205 is completed.

```
.trac 5 1205

IDLE VTN 108 0 00 25   MARP
```

c) SIP Trunk monitoring (LD 32)
Place an inbound call from the PSTN (6139675258) to the CS1000 (7162611205). Then check the SIP Trunk status by using LD 32.

```
>ld 32
NPR000
.stat 104 1
063 UNIT(S) IDLE
001 UNIT(S) BUSY
000 UNIT(S) DSBL
000 UNIT(S) MBSY
.
```

The following is an example after the call is completed; the BUSY trunk changes its state to IDLE.

```
.stat 104 1
064 UNIT(S) IDLE
000 UNIT(S) BUSY
000 UNIT(S) DSBL
000 UNIT(S) MBSY
.
```

# 10. Conclusion

These Application Notes describe the configuration necessary to connect an Avaya Communication Server 1000 7.5, an Avaya Aura® Session Manager 6.1 and an Acme Packet Session Border Controller 6.2 to Level 3 SIP Trunking Service. Level 3 SIP Trunking Service is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. Level 3 SIP Trunking Service provides a flexible, cost-saving alternative to traditional analog and ISDN-PRI trunks.

All of the test cases have been executed. Despite the number of observations and limitations seen during testing as noted in **Section 2.2**, the test results met the objectives outlined in **Section 2.1**. The Level 3 SIP Trunking Service is considered **compliant** with Avaya Communication Server 1000 7.5, Avaya Aura® Session Manager 6.1 and Acme Packet Session Border Controller 6.2.

# 11. References

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


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

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

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

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

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

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

[7] *Installing and Configuring Avaya Aura® System Platform,* Release 6.03, February 2011.

[8] *Administering Avaya Aura® System Platform,* Release 6, June 2010.

[9] *Installing and Upgrading Avaya Aura® System Manager*, Release 6.1, November 2010.

[10] *Installing and Configuring Avaya Aura® Session Manager*, Release 6.1, April 2011, Number 03-603473.

[11] *Administering Avaya Aura® Session Manager*, Release 6.1, May 2011, Document Number 03-603324.

[12] *Acme Packet Net-Net® EMS User Guide,* Release Version 4.1.

[13] *RFC 3261 SIP: Session Initiation Protocol*, http://www.ietf.org/

[14] *RFC 3262, Reliability of Provisional Responses in the Session Initiation Protocol (SIP)* http://www.ietf.org/

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


Product documentation for Level 3 SIP Trunking Service is available from Level 3.

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