Application Notes for Configuring Level 3 SIP Trunking with the Avaya Communication Server 1000 Release 7.5, Avaya Aura® Session Manager Release 6.1 and Avaya Aura® Session Border Controller Release 6.0.2 - 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 Avaya Aura® Session Border Controller 6.2, Avaya Aura® Session Manager 6.1, Avaya Communication Server 1000 7.5 and various Avaya endpoints. During the interoperability testing, Avaya Communication Server 1000 was able to interoperate with Level 3 via SIP trunk. This test was performed to verify SIP trunk features including basic call, call forward (all calls, busy, no answer), call transfer (blind and consult), conference and voice mail. The calls are placed in both directions with various set types. This documented solution does not extend to configurations without Avaya Aura® Session Border Controller or Avaya Aura® Session Manager.

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

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1. Introduction

This document provides a typical network configuration deployment of the Avaya Communication Server 1000E (hereafter referred to as CS1000E) and the Level 3 SIP Trunking (hereafter referred to as Level 3). During the interoperability testing, all SIP trunk applicable feature test cases were executed to ensure the interoperability between the Level 3 system and the Avaya CS1000E 7.5, Avaya Aura® Session Manager Release 6.1 and Avaya Aura® Session Border Controller Release 6.2 system.

2. General Test Approach and Test Results

The CS1000E system release 7.5 was connected to an Avaya Aura® Session Border Controller (hearafter referred to as the Avaya Aura® SBCvia the Avaya Aura® Session Manager (hereafter referred to as Session Manager).Then the Avaya Aura® SBC was connected to the Level 3 system via SIP trunk. Various call types were made from the CS1000E to Level 3 and vice versa to ensure the interoperability between the systems.

Level 3 is a member of the Avaya DevConnect Service Provider program. DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The focus of this testing is to verify that CS1000E release 7.5 can interoperate with the Level 3. The following interoperability areas were covered.

- General call processing between CS1000E and Level 3 systems including:
  - Codec (G.711 u-law and G.729/ ptime 20ms/ VAD disabled).
  - Hold/Retrieve on both ends.
  - Music On Hold.
  - CLID displays.
  - Ring-back tone.
  - Speech paths.
  - Dialing plan support.
  - Advanced features (Call on Mute, Call Park, and Call Waiting).
  - Abandoned Call.
- Call redirection verification: all supported methods (blind transfer, consultative transfer, call forward, and conference) including CLID. Call redirection was performed from both ends.
- RFC2833/DTMF on both directions.
- SIP Transport UDP with Digest Authentication.
- SIP Digest Authentication.
- Thru dialing via PBX Call Pilot.
- Voice Mail Server CallPilot (hosted on CS1000E).
- Fax Transmission: the fax call was transmitted from both ends with codec T.38.
- Early Media Transmission.

2.2. Test Results

The general test approach was to configure a simulated enterprise site using the CS1000E, Session Manager and the Avaya Aura® SBC to connect to the Level 3 SIP Trunking service.

This configuration, shown in **Figure 3.1**, was used to exercise the features and functionality listed in **Section 2.1**.

Interoperability testing of Level 3 SIP Trunking service with the Avaya SIP-enabled enterprise solution was completed with successful results with the exception of the observations/limitations described in this section.

2.2.1. Blind Transfer

In the default configuration, the CS1000E will not allow a blind transfer to be executed if the SIP trunk service, in this case Level 3, does not support the SIP UPDATE method. With the installation of plug-in 501 on the CS1000E, the blind transfer will be allowed and the call will be completed. The limitation of this plugin is that no ringback is provided to the originator of the call for the duration that the destination set is ringing. In addition to plug-in 501, it is required that VTRK SU version "cs1000-vtrk-7.50.17.16-15.i386.000.ntl", as detailed in **Section 4**, or higher be used on all SSG signaling servers to ensure proper operation of the blind transfer feature.

Example scenario:

Assume a call is active between a CS1000E telephone user and a PSTN user "A". To allow the CS1000E user to transfer the call using the Level 3 SIP Trunk service to another PSTN user B before user B has answered the call, CS1000E plug-in 501 must be enabled as shown in **Section 4**. While plug-in 501 will allow the CS1000E user to complete the transfer operation, user A will not hear ring back tone while user B is ringing in this case. PSTN users A and B will have two-way talk path once user B answers.

2.2.2. History-Info and Diversion Headers

The Level 3 service does not support SIP History-Info headers. The Level 3 service requires that SIP Diversion Header be sent for certain redirected calls (e.g. Call Forward). Session Manager is used to convert the History Info header into the Diversion Header by the use of the adaptation "DiversionTypeAdapter" for these types of calls. For all other calls, the Avaya Aura® SBC will strip off History-Info headers.

2.2.3. SIP Header Optimization

SIP header rules were implemented on the Avaya Aura® SBC to streamline the SIP header and remove any unnecessary parts. The following headers were removed: X_nt_e_164_clid, Alert_Info, P-Location, P-Site, Alert-Info, History-Info, x-nt-corr-id and P-Asserted-Identity. Also the multipart MIME SDP, which included the x-nt-mcdn-frag-hex, x-nt-epid-frag and x-nt-inforeq/8000, was stripped out. These particular headers and MIME have no real use in the service provider network and their presence may add unnecessary confusion.

2.2.4. G.711 Fax

G.711 fax is not supported in the reference configuration. T.38 faxing is supported, as is Group 3 and Super Group 3 fax. Fax speeds to 14400 bps are supported in the configuration tested.

2.2.5. Emergency 911/E911 Services Limitations and Restrictions

911/E911 test calls were not made during the testing of this solution. Although Level 3 provides 911/E911 calling capabilities, Level 3 does not warrant or represent that the equipment and software reviewed in this customer configuration guide will properly operate to complete 911/E911 calls; therefore, it is the customer's responsibility to ensure proper operation with the equipment/software vendor.

2.2.6. Toll Free

Inbound toll free calls were not tested as part of this solution.

2.3. Support

For technical support on Level 3 system, please contact Level 3 technical support at:

- Toll Free: 1-877-4LEVEL3 (1-877-453-8353)
- http://www.level-3.voip.com/en/contact-us/

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. Selecting the Support Contact Options link followed by Maintenance Support provides the worldwide support directory for Avaya Global Services. Specific numbers are provided for both customers and partners based on the specific type of support or consultation services needed. Some services may require specific Avaya service support agreements. Alternatively, in the United States, (866) GO-AVAYA (866-462-8292) provides access to overall sales and service support menus.

3. Reference Configuration

Figure 3.1 illustrates the test configuration used during the compliance testing event between the CS1000E and Level 3.

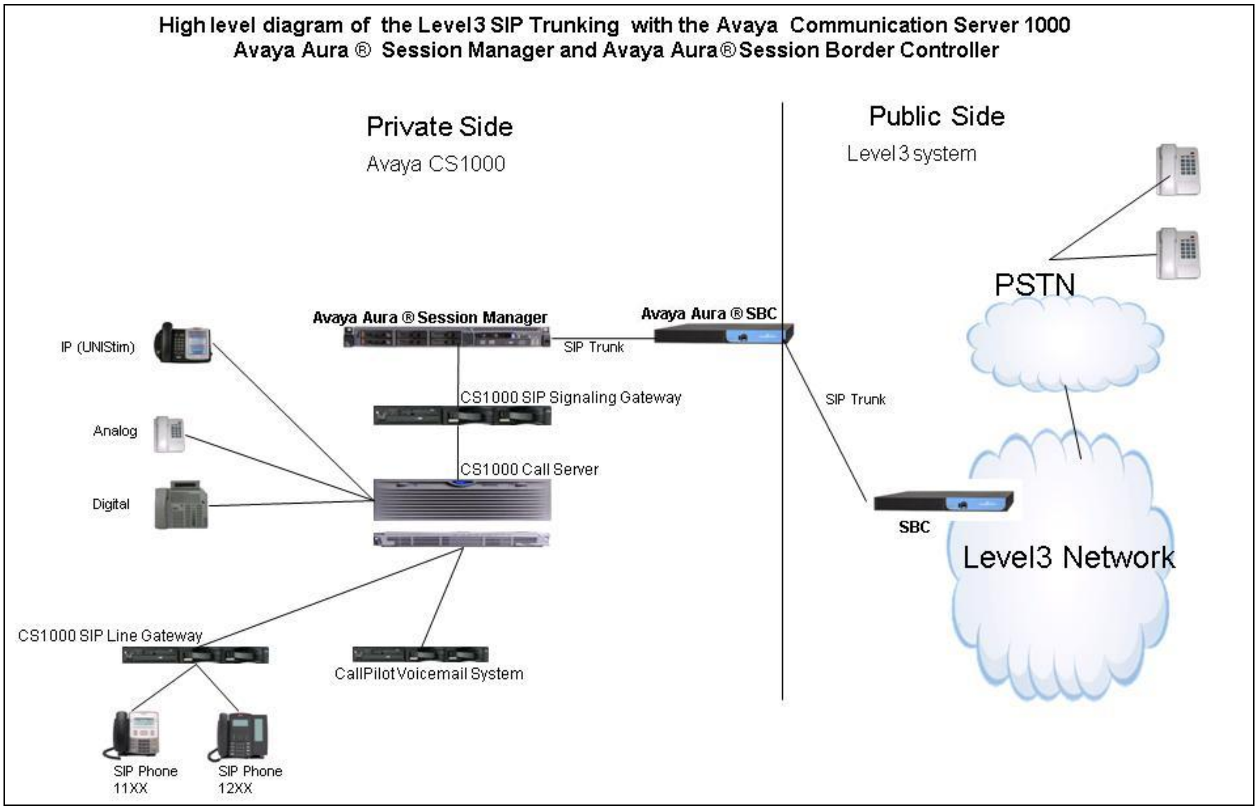


Figure 3.1 Network Diagram for Avaya CS1000E – Level 3

The following assumptions were made for this lab test configuration.

1. CS1000E R7.5, Session Manager 6.1 and Avaya Aura® SBC software implemented with all the latest patches.
2. Level 3 provides support to setup, configure, and troubleshoot on carrier switch for the duration of the testing.

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

1. Calls were checked for the correct call progress tones and cadences.
2. During the ringing state, the ring back tone and destination ringing were checked.
3. Calls were checked in both hands-free and handset mode due to internal Avaya requirement.
4. Calls were checked for speech path in both directions using spoken words to ensure clarity of speech.

5. The display(s) of the sets/clients involved were checked for consistent and expected CLID, name 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.

The Fully Qualified Domain Names and IP addressing specified in these Application Notes apply only to the reference configuration shown in Figure 3.1. Avaya uses a combination of FQDNs and IP addresses, the Level 3 network is IP address based.
For confidentiality purposes, the IP addresses in these Application notes have been modified to show 111.x.x.x for Avaya internal addresses, 222.x.x.x for Avaya external address and 333.x.x.x for Level 3 external address. Level 3 customers will use their own FQDNs and IP addresses as required

4. Equipment and Software Validated

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

Avaya system:

Avaya IP Telephony Solution Components	
Equipment / Software	Release / Version
Avaya Communication Server 1000E running on CP+DC server as co-resident configuration	• Call Server: 7.50 .17 GA (CoRes) Service Pack: 7.50.17_20120110 • SSG Server: 7.50.17 GA • SLG Server: 7.50.17 GA
Communication Server 1000E Media Gateway	CSP Version: MGCC CD02 MSP Version: MGCM AB01 APP Version: MGCA BA15 FPGA Version: MGCF AA19 BOOT Version: MGCB BA15 DSP1 Version: DSP4 AB01 BCSP Version: MGCC CD01
Avaya Aura® Session Manager	6.1.1.0.611023
Avaya Aura® Session Border Controler	6.0.2.0.3
Avaya i2001 IP Telephone (UNIStim)	0604DCN
Avaya 2050 IP Softphone (UNIStim)	4.2.0062
Avaya 1140 IP Telephone (SIP)	04.03.12.00
Avaya M3904 (Digital)	n/a
Avaya 6210 Analog Telephone	n/a
Level 3 SIP Trunking Solution Components	
Equipment / Software	Release / Version
Level 3 Enterprise Edge	Version 1

Additional software and patch lineup for the configuration and active patch list are listed as below.
Call Server: 7.50 Q GA plus latest DEPLIST – Issue: 01 Release: x2107.50, 2011-07-19 11:40:08 (est)
SSG Server: 7.50.17 GA plus latest Service_Pack_Linux_7.50_17.16-1.i386.000.ntl
SLG Server: 7.50.17 GA plus latest Service_Pack_Linux_7.50_17.16-1.i386.000.ntl

Note: It is required that VTRK SU version "cs1000-vtrk-7.50.17.16-15.i386.000.ntl" or higher be used on all SSG signaling servers to ensure proper operation of the blind transfer feature. The pstat command shown below can be used to verify what version of VTRK SU is installed. If a new version is required, download the newest Linux 7.50 Service Pack and install using the standard patch process (not described in this document).

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 :  2011-07-19 11:40:08 (est)
        Number of patches: 60
        Patches Loaded: 60
        Patches In-service: 60
```

The output of "**pstat**" command on SSG Server:

```
[admin@car1-sps-ucm ~]$ pstat
Product Release: 7.50.17.00
In system patches: 0

In System service updates: 12
PATCH#  IN_SERVICE   DATE      SPECINS   REMOVABLE   NAME
0       Yes         27/04/11   NO       YES        cs1000-sps-7.50.17-01.i386.000
1       Yes         27/04/11   NO       YES        cs1000-baseWeb-7.50.17.01-1.i386.000
2       Yes         27/04/11   NO       YES        cs1000-shared-pbx-7.50.17-01.i386.000
3       Yes         27/04/11   NO       YES        cs1000-dbcom-7.50.17-02.i386.000
4       Yes         29/08/11   NO       YES        cs1000-vtrk-7.50.17.16-15.i386.000
11      Yes         25/08/11   NO       YES        cs1000-linuxbase-7.50.17.16-1.i386.000
12      Yes         25/08/11   NO       YES        cs1000-dmWeb-7.50.17.16-1.i386.000
13      Yes         25/08/11   NO       YES        cs1000-emWeb_6-0-7.50.17.16-6.i386.000
14      Yes         25/08/11   NO       YES        cs1000-tps-7.50.17.16-4.i386.000
15      Yes         25/08/11   YES      YES        cs1000-Jboss-Quantum-7.50.17.16-4.i386.000
16      Yes         25/08/11   NO       YES        cs1000-patchWeb-7.50.17.16-1.i386.000
17      Yes         25/08/11   NO       YES        cs1000-bcc-7.50.17.16-13.i386.000
```

The plug-in list can be displayed with the plp (plug-in print) command as shown below. Plug-ins come preinstalled and are delivered with every software load. If plug-in 501 is not activated, it can be enabled using the ple command, also shown below.

```
>
PDT login on /pty/ptty00.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>
```

```
pdt> ple 501

PLUG-IN 501  IS ENABLED
```

5. Avaya Communication Server 1000 Configuration

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

The below procedures describe the configuration details of CS1000E with a SIP trunk to the Level 3 system.

5.1. Login to CS1000E System

5.1.1. Login Unified Communications Management and Element Manager

a) Open an instance of a web browser and connect to the Unified Communications Management (UCM) GUI at the following address: http://<UCM IP address> as shown in **Figure** 5.1. Log in using an appropriate Username and Password.

Figure 5.1 Login Unified Communications Management

b) The **Unified Communications Management** screen is displayed. Click on the **Element Name** of the CS1000E Element as highlighted in red box as shown in **Figure** 5.2.

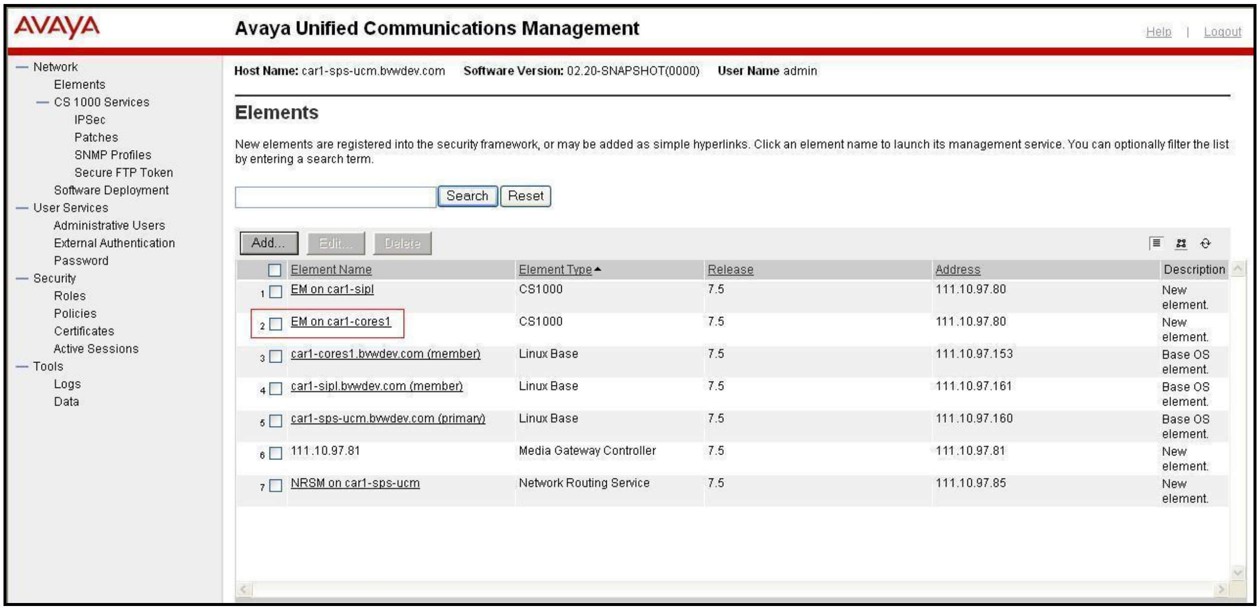


Figure 5.2 Unified Communications Management

c) The CS1000E Element Manager (EM) **System Overview** page is displayed as shown in **Figure 5.3**, this is the main Element Manager screen from which all other menus can be launched.

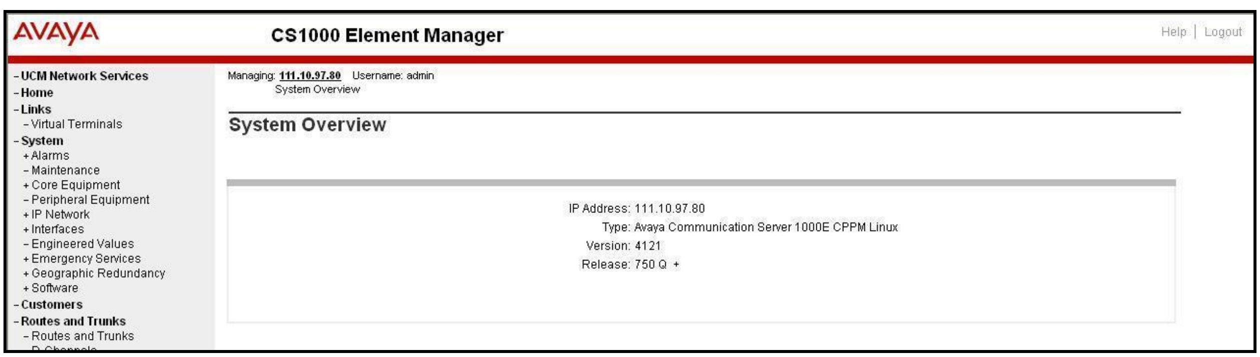


Figure 5.3 Element Manager System Overview

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

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

```
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@111.10.97.80's password:
Last login: Mon Jul 18 11:01:44 2011 from 135.20.233.246
[admin@car1-cores1 ~]$ cslogin

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

TTY 09 SCH MTC BUG   11:38
OVL111 IDLE   0
>login admin
PASS?
.
TTY #09 LOGGED IN ADMIN 11:3
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.
9  18/7/2011

>
SRPT4619 WARNING: Last Archive Procedure had failed
        No archives were completed since May 13 14:59:00 2011


OVL000
>
```

5.2. Administer a Node IP Telephony

This section describes how to configure a Node IP Telephony on the CS1000E.

5.2.1. Obtain Node IP address

These Application Notes assume that the basic configuration has already been administered and that a Node has already been created. This section describes the steps for configuring a Node (Node ID 1000) in CS1000E IP network to work with the Level 3 system. For further information on Avaya Communications Server 1000, please consult reference in **Additional References**.

a) Select **System** -> **IP Network** -> **Nodes: Servers, Media Cards.**
b) **Figure 5.4** displays **IP Telephony Nodes** page. Then click on the Node ID of your CS1000E Element (e.g. **1000**).

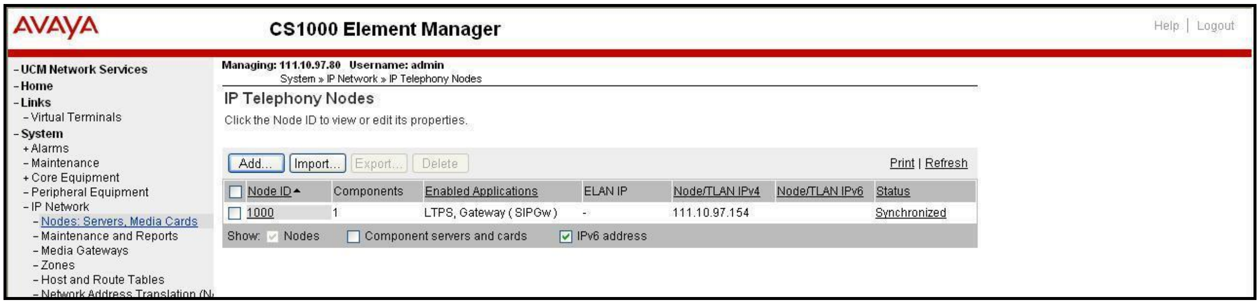


Figure 5.4 IP Telephony Nodes

b) The **Node Details** screen is displayed in
Figure 5.5 with the IP address of the CS1000E node. The **Node IP Address** is a virtual address which corresponds to the TLAN IP address of the Signaling Server, SIP Signaling Gateway. The SIP Signaling Gateway uses this **Node IP Address** to communicate with other components to process the SIP call.

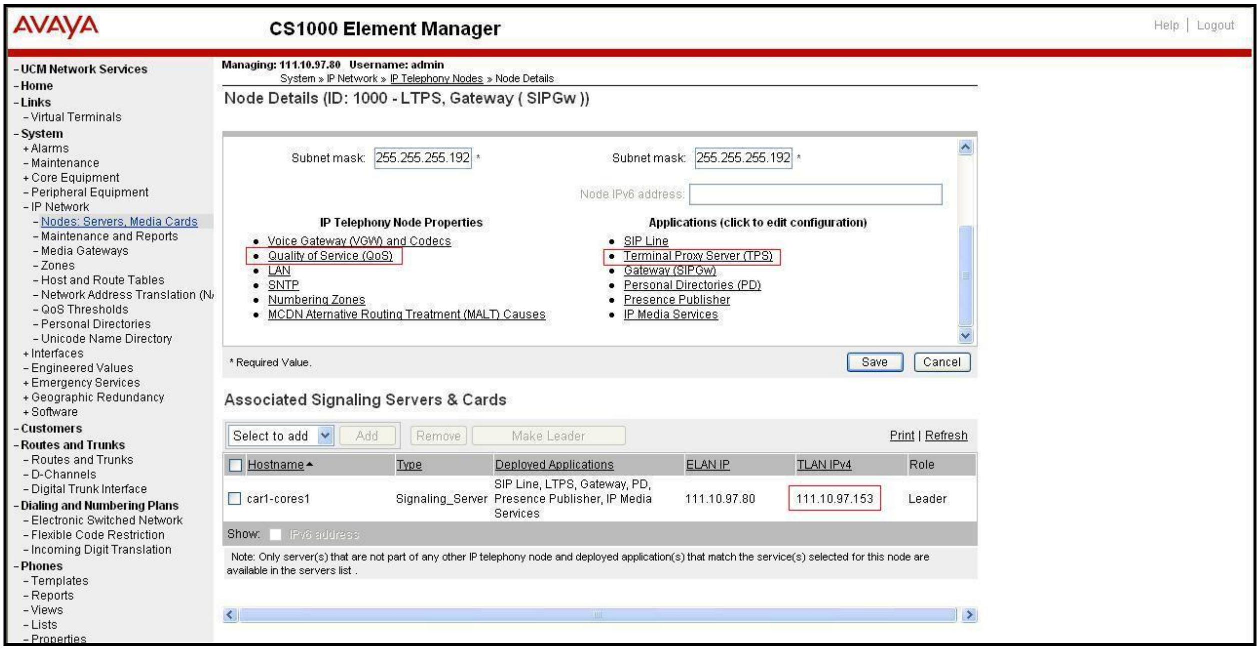


Figure 5.5 Node Details

5.2.2. Administer TPS

c) Continue from **Section 5.2.1**. On the **Node Details** page, select the **Terminal Proxy Server** (**TPS**) link as shown in

Figure 5.6.

d) Check the **UNIStim Line Terminal Proxy Server** check box and then click **Save** as shown in **Figure 5.6**.

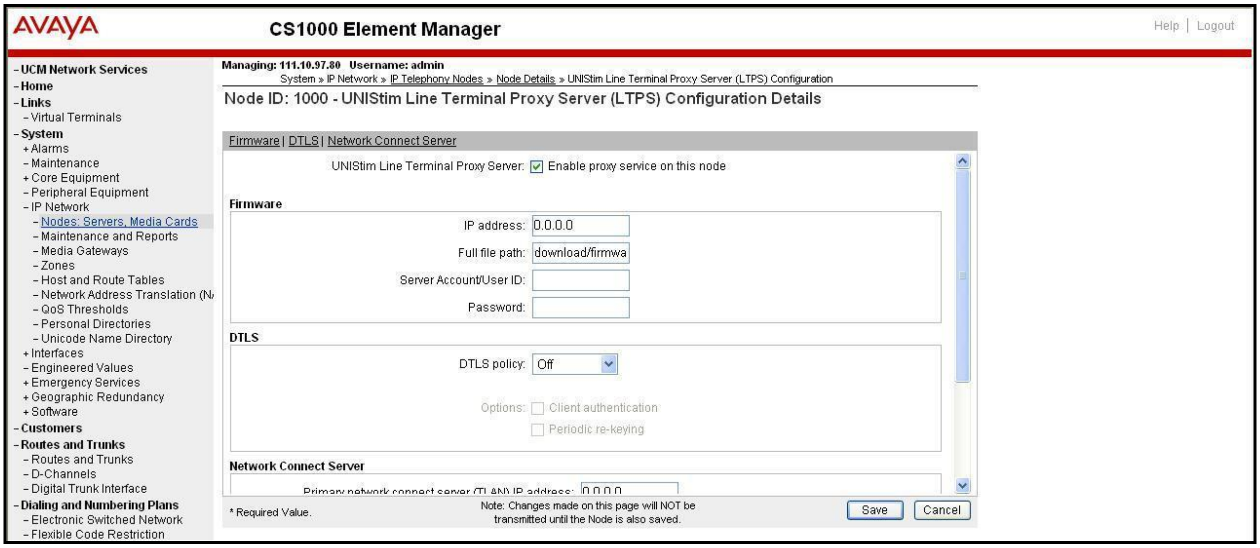


Figure 5.6 TPS Configuration Details

5.2.3. Administer Quality of Service (QoS)

e) Continue from **Section 5.2.1**. On the **Node Details** page, select the **Quality of Service (QoS)** link as shown in
Figure 5.5.

f) The default Diffserv values are as shown in
Figure 5.7. Click the **Save** button.

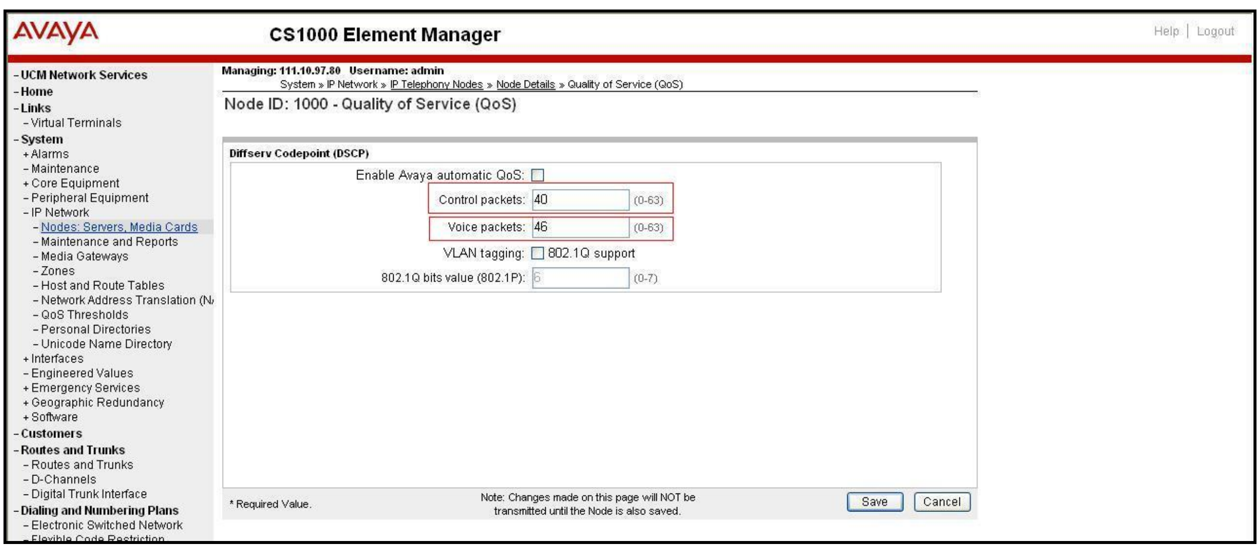


Figure 5.7 QoS Configuration Details

5.2.4. Synchronize the New Configuration

g) Continue from **Section 5.2.3**, return to the **Node Details** page in
Figure 5.5 and click on the **Save** button.
h) The **Node Saved** screen is displayed. Click on the **Transfer Now** (not shown).
i) The **Synchronize Configuration Files** screen is displayed. Check the Signaling Server check box and click on the **Start Sync** (not shown).
j) When the synchronization completes, check the Signaling Server check box and click on the **Restart Applications** (not shown).

5.3. Administer Voice Codec

5.3.1. Enable Voice Codec, Node IP Telephony.

a) Select **IP Network** -> **Nodes: Servers, Media Cards** -> Configuration from the left pane, and in the **IP Telephony Nodes** screen displayed, select the **Node ID** of the CS1000E system. The **Node Details** screen is displayed. (See in **Section 5.2.1** for more detail).
b) On the **Node Details** page as shown in
Figure 5.5, click on **Voice Gateway (VGW) and Codec**.
c) The Level 3 SIP Trunk supports voice codec G.711 and G.729, payload size 20 ms, with VAD disabled.
Figure 5.8 and
Figure 5.9 shows voice codec profile configured on CS1000E with G.729 and G.711, payload size 20ms and VAD disabled.

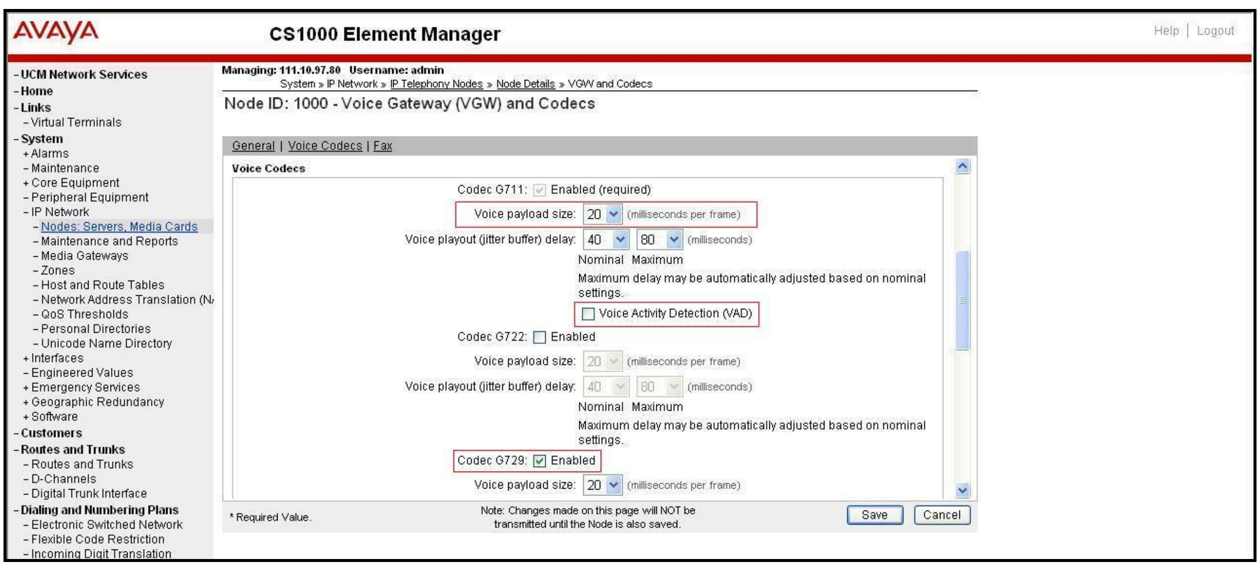


Figure 5.8 Voice Codec G.711 Configuration Details

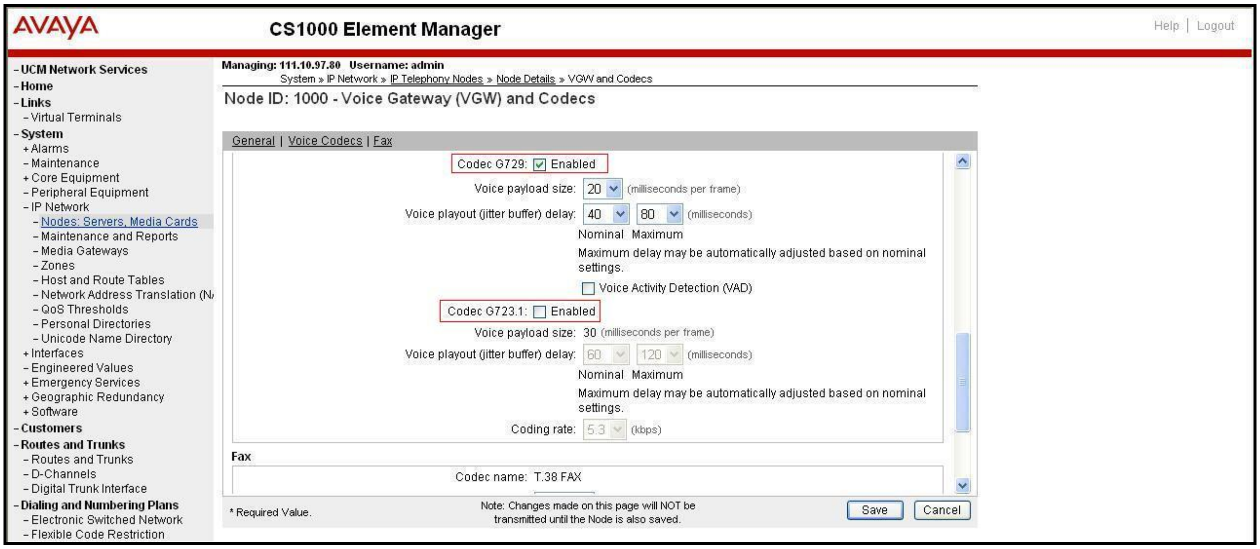


Figure 5.9 Voice Codec G.729 Configuration Details

d) For Fax over IP, Level 3 supports T.38 as default and G.711 as fallback.
Figure 5.10 shows T.38 with payload size 30ms was chosen as default codec for fax.

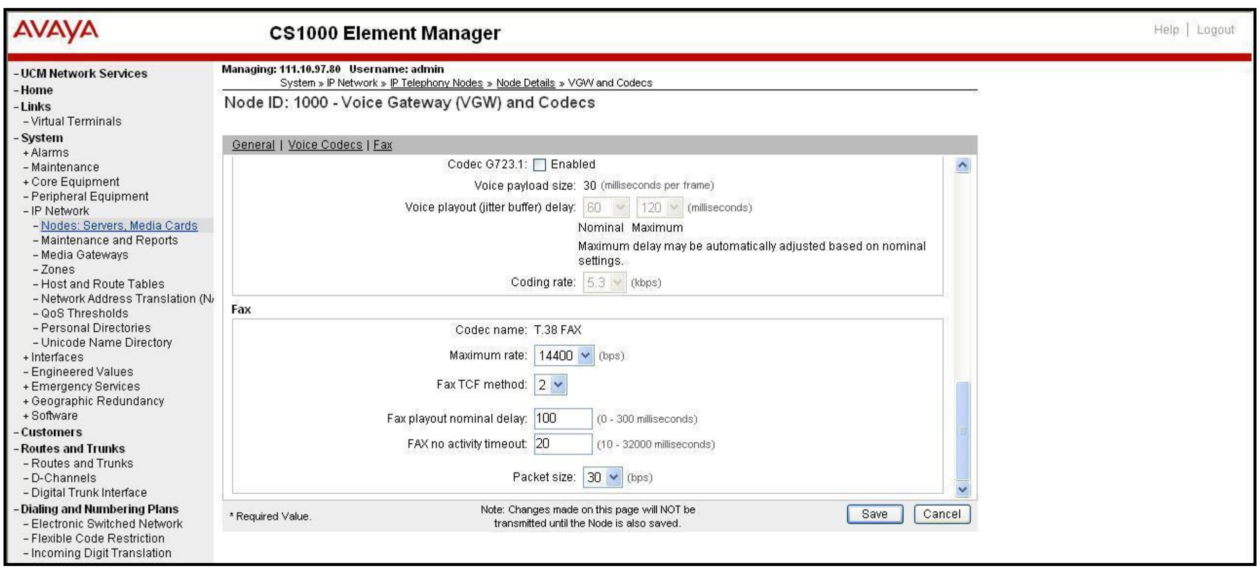


Figure 5.10 Fax Codec T.38 Configuration Details

Figure 5.11 shows **Modem Pass Through** was selected; this configuration enables G.711 as fallback codec for fax.

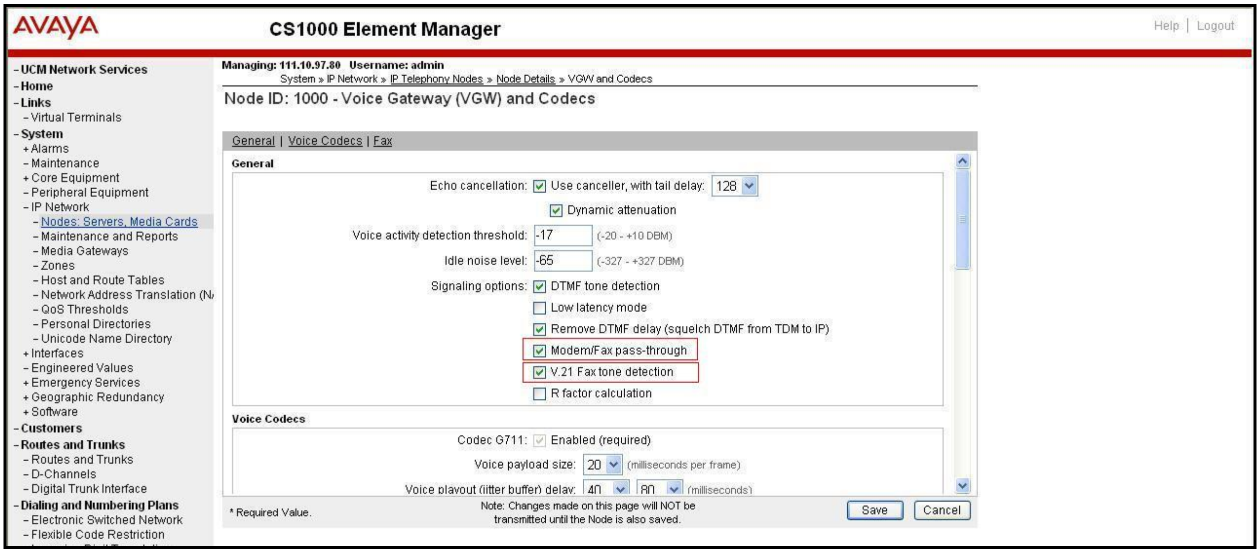


Figure 5.11 Fax Codec G.711 Configuration Details

e) Click **Save**.

f) Synchronize the new configuration (please refer to **Section 5.2.4** for more detail)

5.3.2. Enable Voice Codec on Media Gateways.

CS1000E uses Media Gateways to support traditional analog/ digital phones to make voice call over SIP Trunk. Media Gateways is also needed to support analog terminal to send fax over IP.

a) From the left menu of the Element Manager page in **Figure** 5.12, select **IP Network** -> **Media Gateways** menu item. The Media Gateways page will appear. Click on the corresponding **IPMG** located on the left of the page.

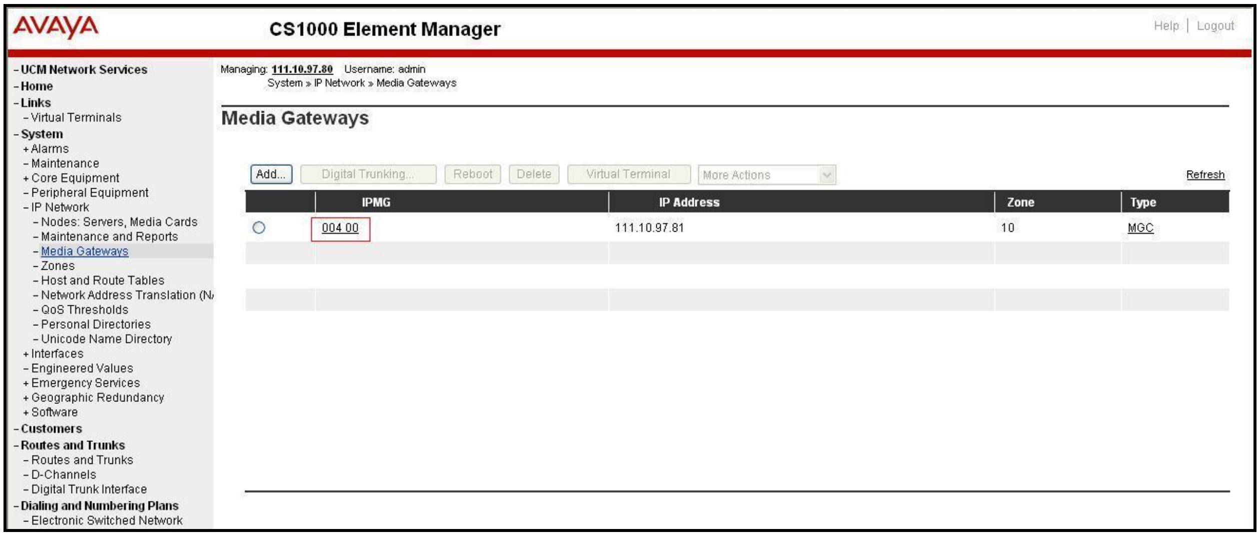


Figure 5.12 Media Gateways Screen

b) The IPMG Property Configuration page displays basic configuration setting for the Media Gateway. Click on the **Next** at the lower right of the page to proceed to the codec settings.

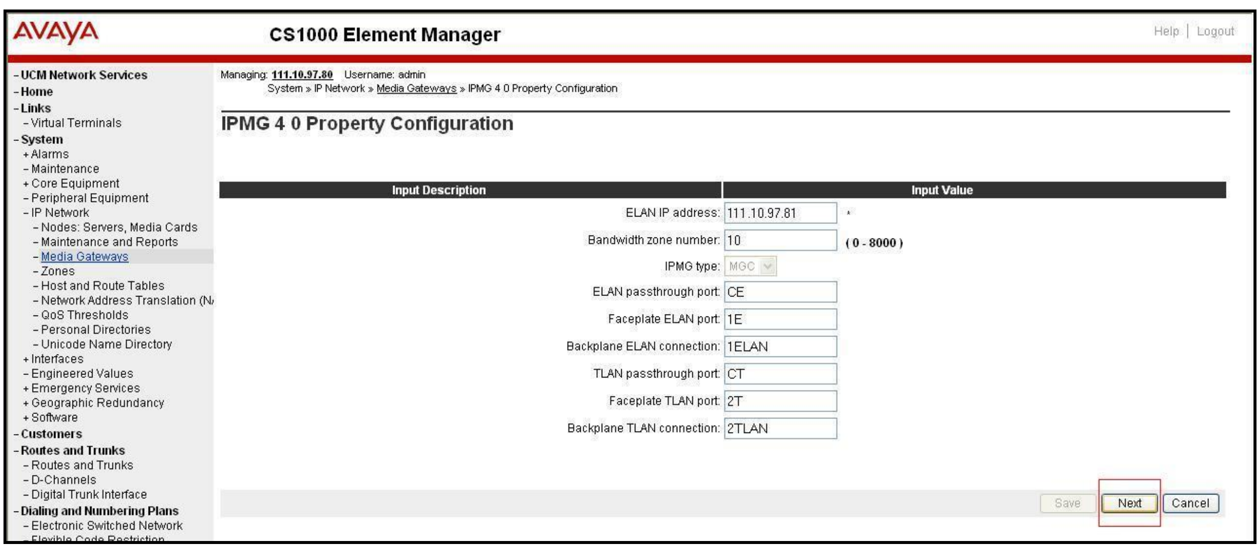


Figure 5.13 IPMG Property Configuration Page

c) The Level 3 SIP Trunk supports voice codecs G.711 and G.729, payload size 20 ms, with VAD disabled.

Figure 5.14 shows configuration for voice codec profile; codec **G711, Voice payload size** 20 and uncheck **VAD**; then check Codec **G729A** checkbox, select **Voice payload size** 20 and uncheck **VAD**.

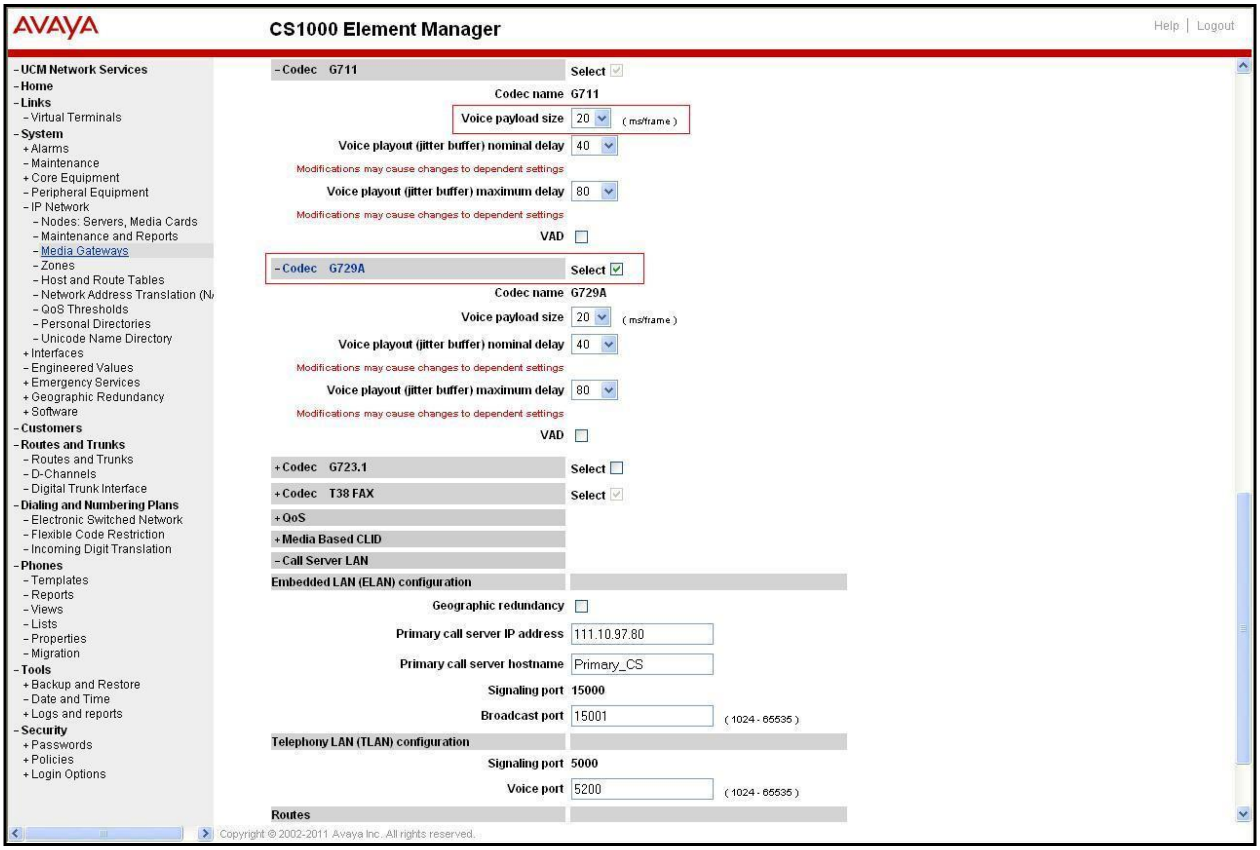


Figure 5.14 Media Gateways G.729 and G.711 Configuration Details

d) For Fax over IP, Level 3 supports T.38 as default and G.711 as fallback.
Figure 5.15 shows T.38 with payload size 30ms was chosen as default codec, and Modem Pass Through was enabled, this configuration enables G.711 as fallback codec for fax.

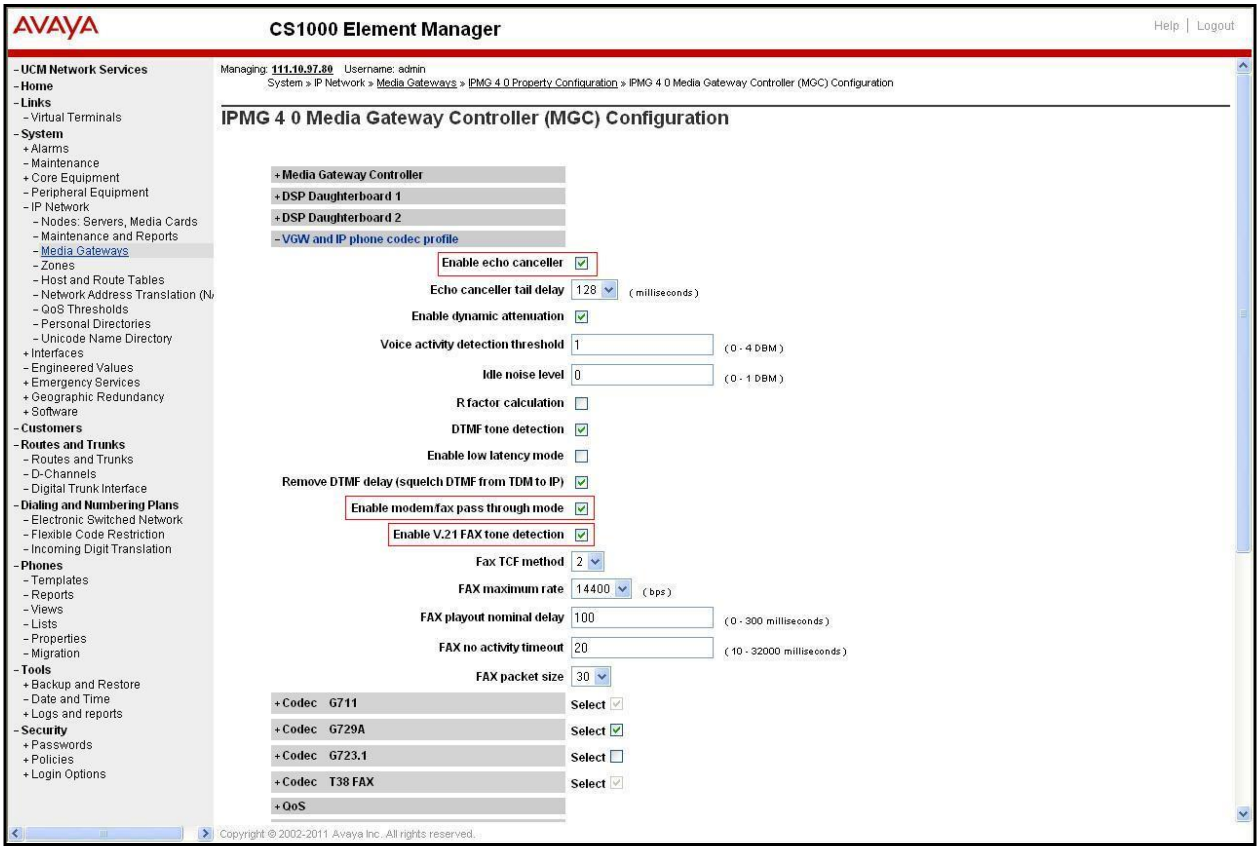


Figure 5.15 Media Gateways T.38 and ModemPassThrough(G.711) Configuration Details

5.4. Administer Zones and Bandwidth

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

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

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

a) Select **IP Network** -> **Zones** configuration from the left pane, click on the **Bandwidth Zones** as shown in
Figure 5.16.

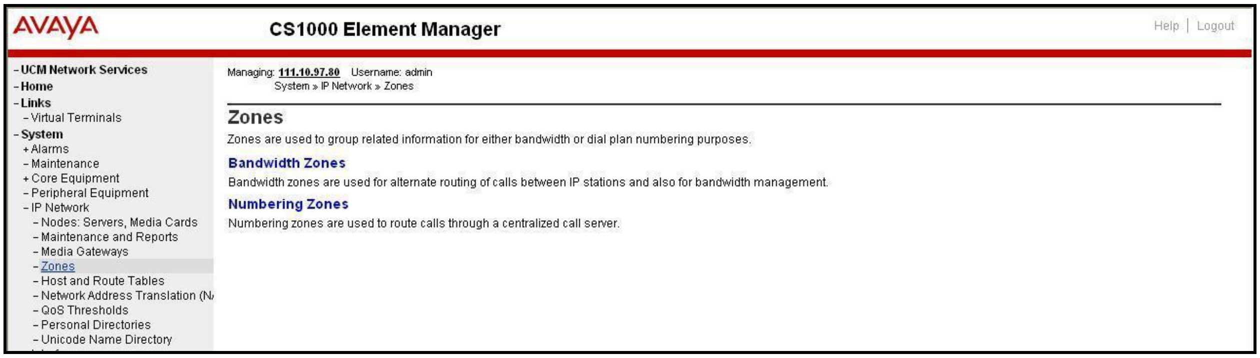


Figure 5.16 Zones Page

b) The **Bandwidth Zones** screen is displayed as shown in
Figure 5.17. Click **Add**.

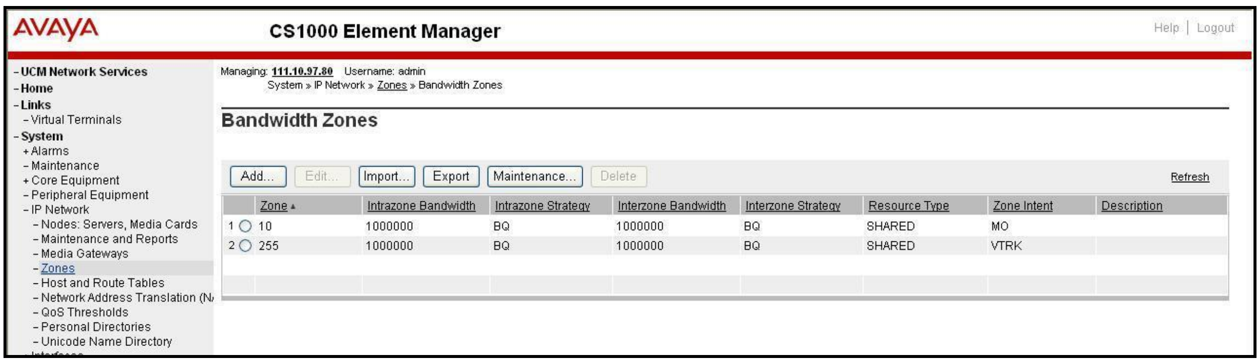


Figure 5.17 Bandwidth Zones

c) Then 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 **Figure** 5.18 and click on the **Submit** button.

- **INTRA_STGY**: bandwidth configuration for local calls.
- **INTER_STGY**: bandwidth configuration for the calls over trunk.
- **BQ**: G711 is first choice and G729 is second choice.
- **BB**: G729 is first choice and G711 is second choice.
- **MO**: is used for IP phones, VGW
- **VTRK**: is used for virtual trunk.

The Level 3 SIP Trunk support is set for G.711 for the initial setup, with G.729 used when necessary for low bandwidth test cases. So the **MO** Zone 10 was configured with **Strategy Best Quality (BQ)**.

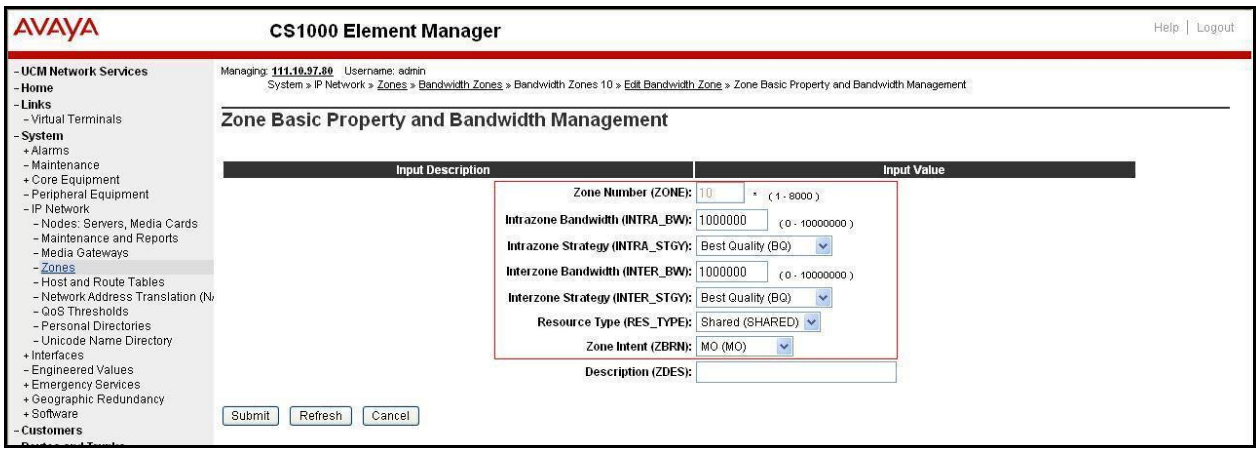


Figure 5.18 Bandwidth Management Configuration Details– IP phone

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

Follow **Section 5.4.1** to create a zone for the virtual trunk. The difference is in **Zone Intent (ZBRN)** field. Select **VTRK** for virtual trunk as shown in

Figure 5.19 and then click on the **Submit** button.

The Level 3 SIP Trunk support G.729 as the first choice, G.711 as fallback. So the **VTRK** Zone 255 was configured with **Strategy Best Quality (BQ)**.

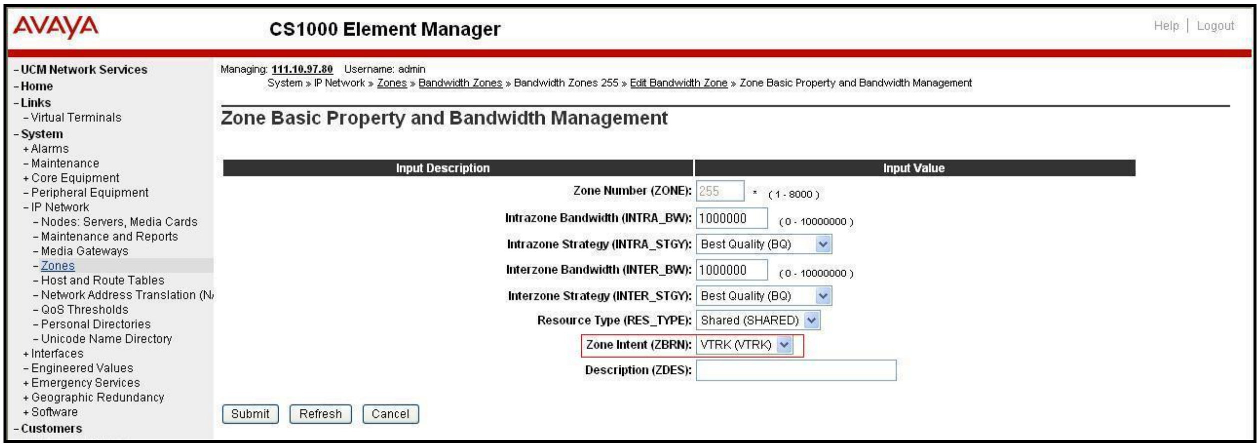


Figure 5.19 Bandwidth Management Configuration Details– Virtual Trunk

5.5. Administer SIP Trunk Gateway

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

5.5.1. Integrated Services Digital Network (ISDN)

a) Select **Customers** in the left pane. The **Customers** screen is displayed. Click on the link associated with the appropriate customer, in this case **00**. The system can support more than one customer with different network settings and options.

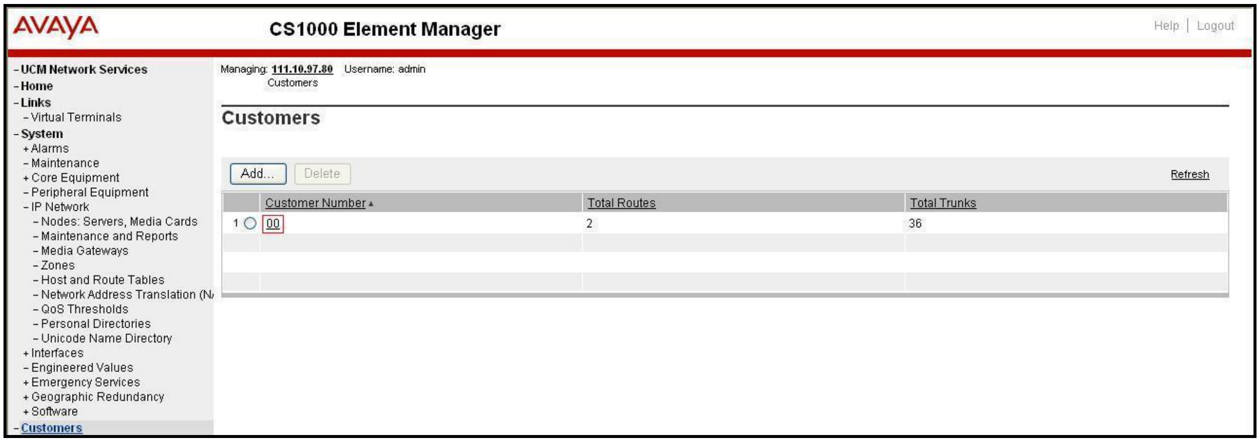


Figure 5.20 Customer Page

b) The **Customer 00 Edit** page will appear. Select the **Feature Packages** option from this page.

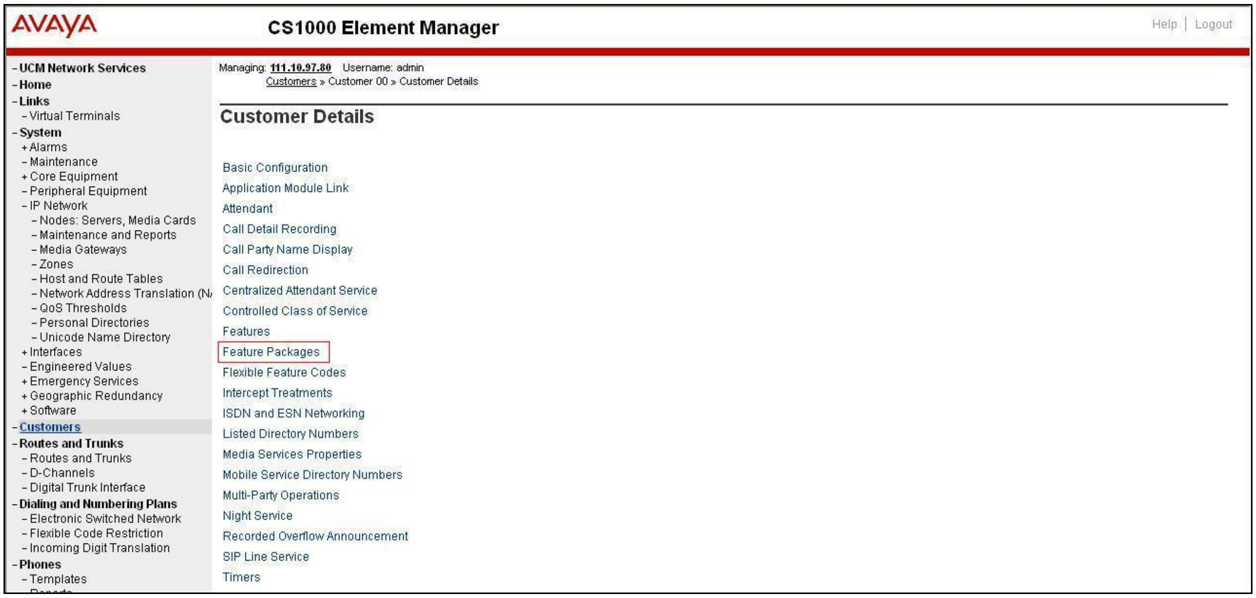


Figure 5.21 Customer Details Page

c) The screen is updated with a list of F**eature Packages** populated. Select **Integrated Services Digital Network** to edit its parameters. The screen is updated with parameters populated below **Integrated Services Digital Network**. Check the **Integrated Services Digital Network** (ISDN) checkbox, and retain the default values for all remaining fields as shown in

Figure 5.22. Scroll down to the bottom of the screen, and click on the **Save** button at the bottom of the page.

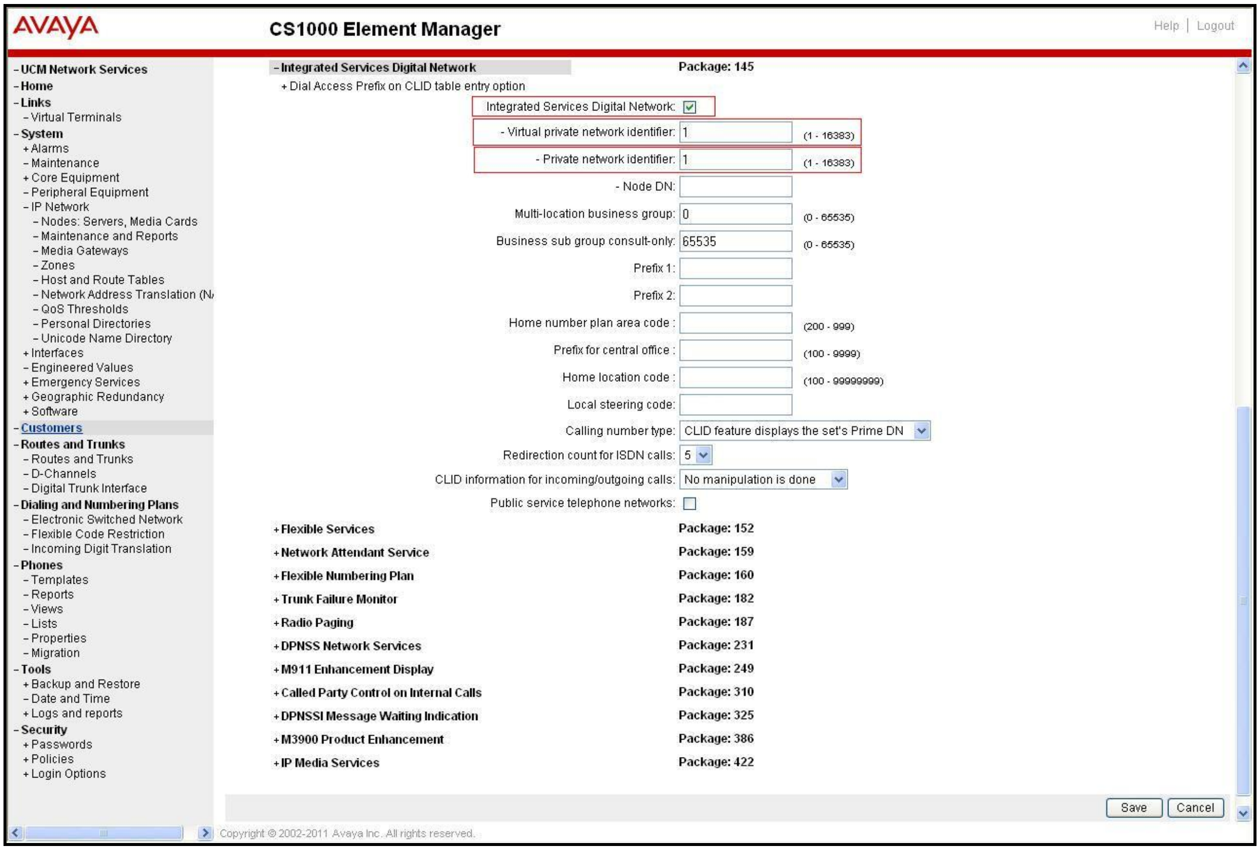


Figure 5.22 Customer – ISDN Configurations

5.5.2. Administer SIP Trunk Gateway to Session Manager

a) Select **IP Network** -> **Nodes: Servers, Media Cards** configuration from the left pane, and in the **IP Telephony Nodes** screen displayed, select the **Node ID** of this CS1000E system. The **Node Details** screen is displayed as shown in
Figure 5.5**, 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 testing values (highlighted in red boxes) for the specified fields, and retain the default values for the remaining fields as shown in
Figure 5.23.
Level 3 implements digest authentication on the SIP Trunk group. Level 3 requires the CS1000E to send the proper **Gateway endpoint name** and MD5 encrypted **Gateway password** in the SIP/INVITE which responds to SIP/401 authentication challenges.
The parameters (highlighted in red boxes) are filled in, which were obtained when user creates a SIP profile in Session Manager (these are shown in **Section6.4**).

- **Vtrk gateway application**: **SIP Gateway (SIPGw)**
- **SIP domain name**: level-3.voip.com
- **Local SIP port**: 5060
- **Gateway endpoint name**: 1-23Q-3413 (the endpoint name as defined by Level 3)
- **Gateway password**: the password defined by Level3

- **Application node ID**: 1000 (this should match the Node ID configured in **Section 4.2.1**)

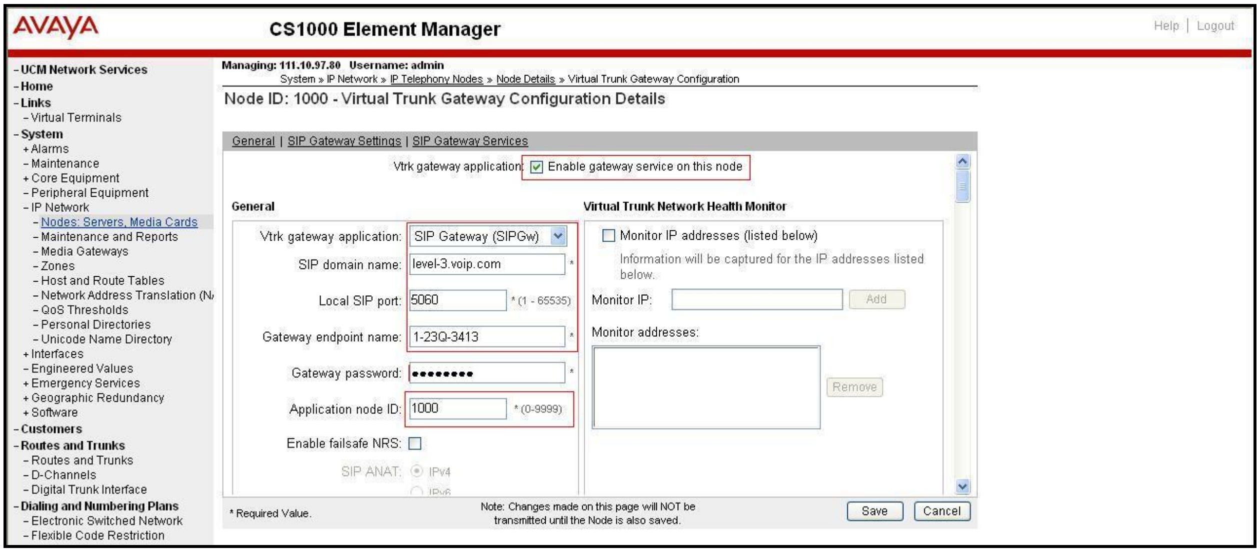


Figure 5.23 Virtual Trunk Gateway Configuration Details Page 1

d) Click on the **SIP Gateway Settings** tab, under **Proxy or Redirect Server**, enter the IP address of Session Manager (highlighted in red boxes) for the specified fields, and retain the default values for the remaining fields as shown in
Figure 5.24.

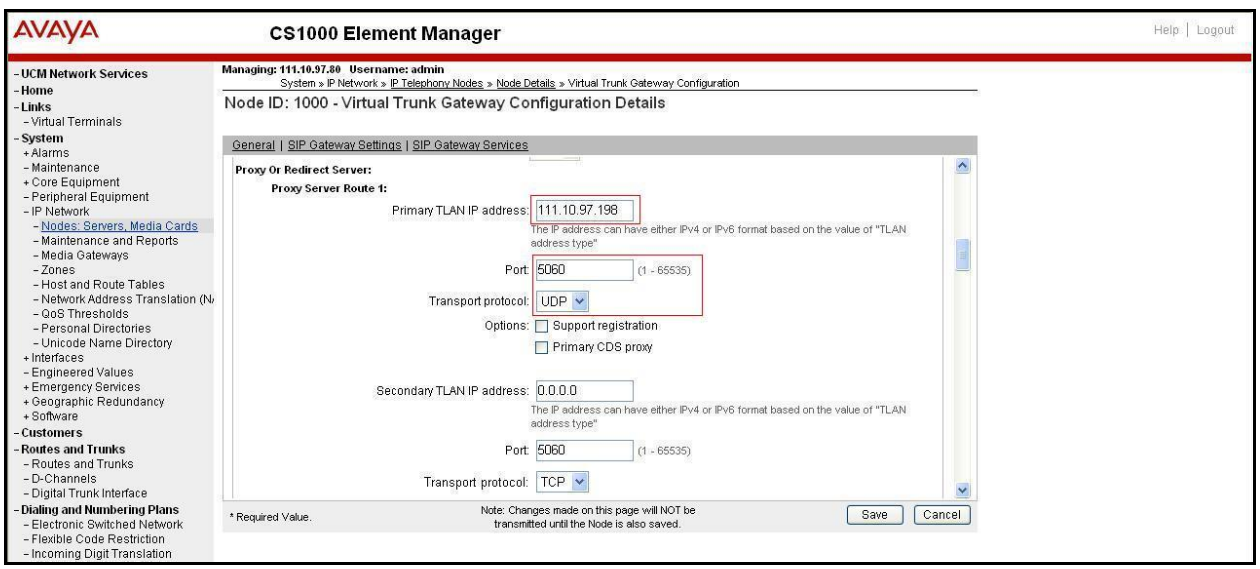


Figure 5.24 Virtual Trunk Gateway Configuration Details Page 2

e) On the same page as shown in
Figure 5.25, scroll down to the **SIP URI Map** section.

Under the Public E.164 Domain Names, for:
- **National**: leave this SIP URI field as blank
- **Subscriber**: leave this SIP URI field as blank
- **Special Number**: leave this SIP URI field as blank
- **Unknown**: leave this SIP URI field as blank

Under the Private E.164 Domain Names, for:
- **UDP**: leave this SIP URI field as blank
- **CDP**: leave this SIP URI field as blank
- **Special Number**: leave this SIP URI field as blank
- **Vacant number**: leave this SIP URI field as blank
- **Unknown**: leave this SIP URI field as blank

Note: These fields are blank in correspondence with the Avaya DevConnect lab configuration; it is possible that customer installations will have SIP URI configured here.

Then click on the **Save** button.

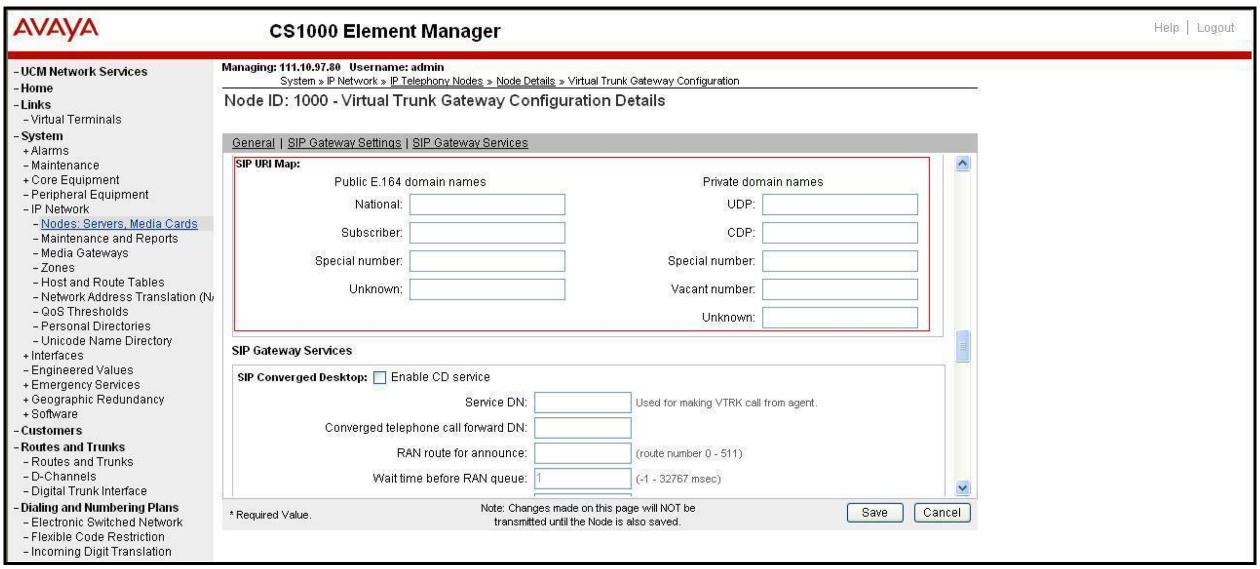


Figure 5.25 Virtual Trunk Gateway Configuration Details Page 3

5.5.3. Administer Virtual D-Channel

a) Select **Routes and Trunks** -> **D-Channels** from the left pane to display the **D-Channels** screen. In the **Choose a D-Channel Number** field, select an available D-channel from the drop-down list as shown in
Figure 5.26. Click on **to Add** button**.**

Figure 5.26 D-Channels

b) The D-Channels 100 Property Configuration screen is displayed next as shown in **Figure** 5.27. Enter the following values for the specified fields, and retain the default values for the remaining fields.

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

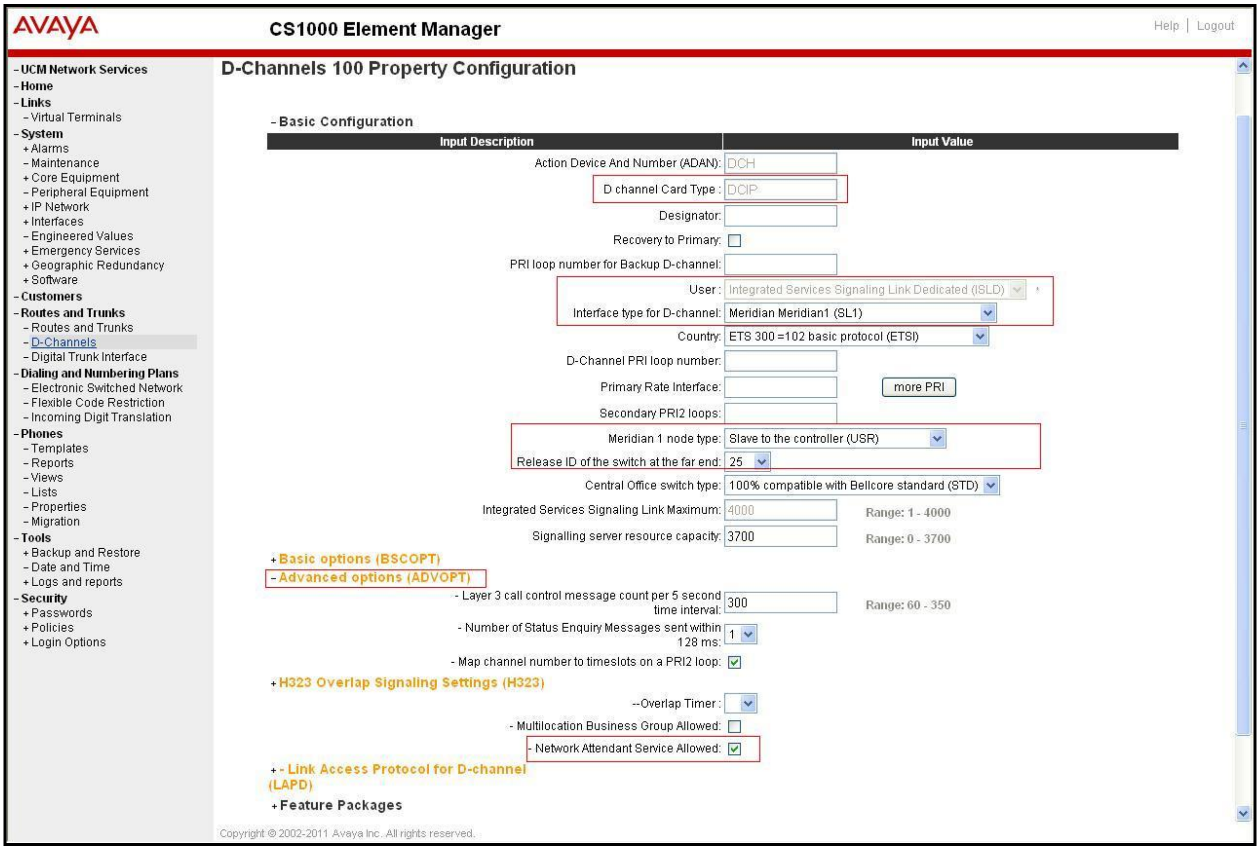


Figure 5.27 D-Channels Configuration Details

c) Click on the **Basic Options** and click on the **Edit** button at the **Remote Capabilities** (**RCAP**) attribute as shown in

Figure 5.28. The **Remote Capabilities Configuration** page will appear. Then check on the **ND2** and the **MWI** (if PSTN mailboxes are present on the CS1000E Call Pilot) checkboxes as shown in

Figure 5.29.

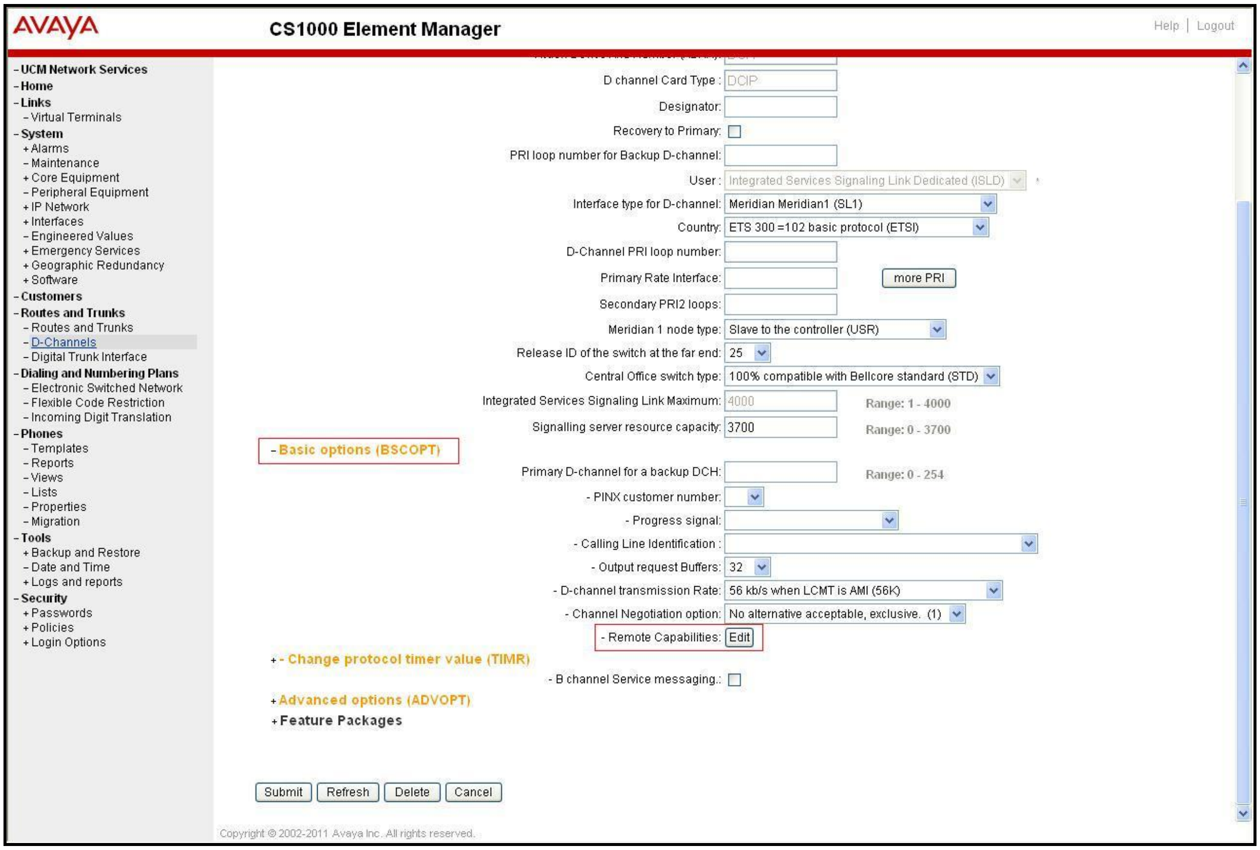


Figure 5.28 D-Channels Configuration Details

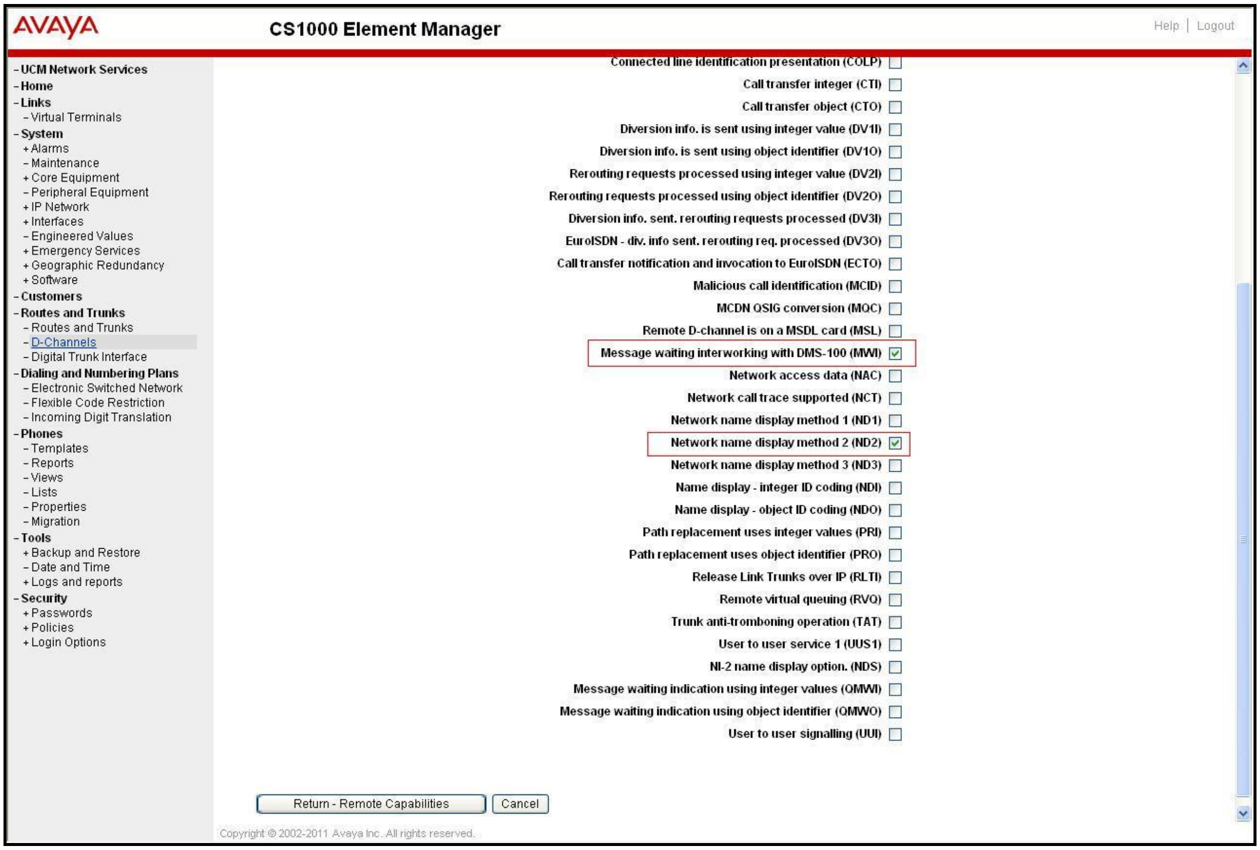


Figure 5.29 Remote Capabilities Configuration Details

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

5.5.4. Administer Virtual Super-Loop

Select **System** -> **Core Equipments** -> **Superloops** from the left pane to display the **Superloops** screen. If the Superloop does not exist, please click "**Add**" button to create a new one as shown in
Figure 5.30. In this example, Superloop 100 is being added and used.

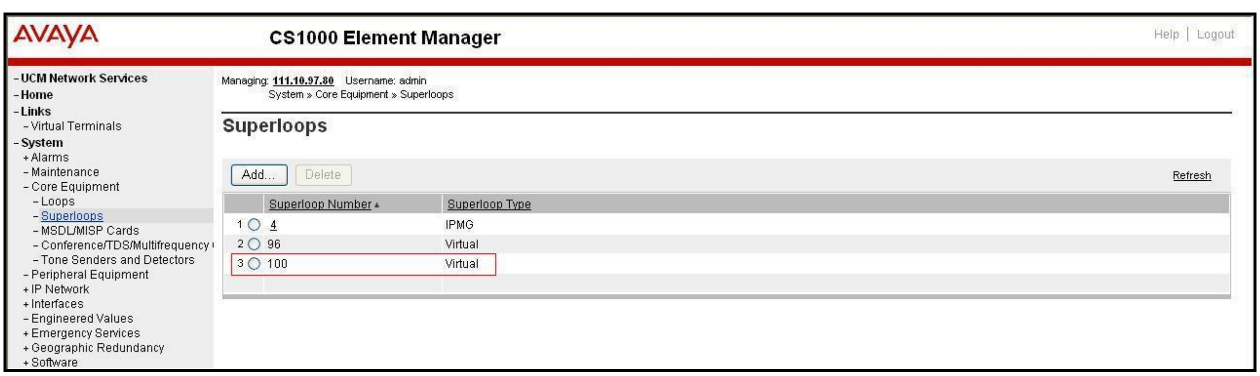


Figure 5.30 Administer Virtual Super-Loop

5.5.5. Enable Music for Customer Data Block

a) Select **Customers** in the left pane. The **Customers** screen is displayed. Click on the link associated with the appropriate customer, in this case **00**. The system can support more than one customer with different network settings and options. The **Customer 00 Edit** page will appear (not shown). Select the **Feature Packages** option from this page.
b) The screen is updated with a list of F**eature Packages** populated. Select **Enhanced Music** to edit its parameters. Check to enable music for Customer 00, define music route 1 as show in the red box of
Figure 5.31. The CS1000E system has been pre-configured with music route 1.
c) Scroll down to the bottom of the screen, and click on the **Save** button at the bottom of the page.

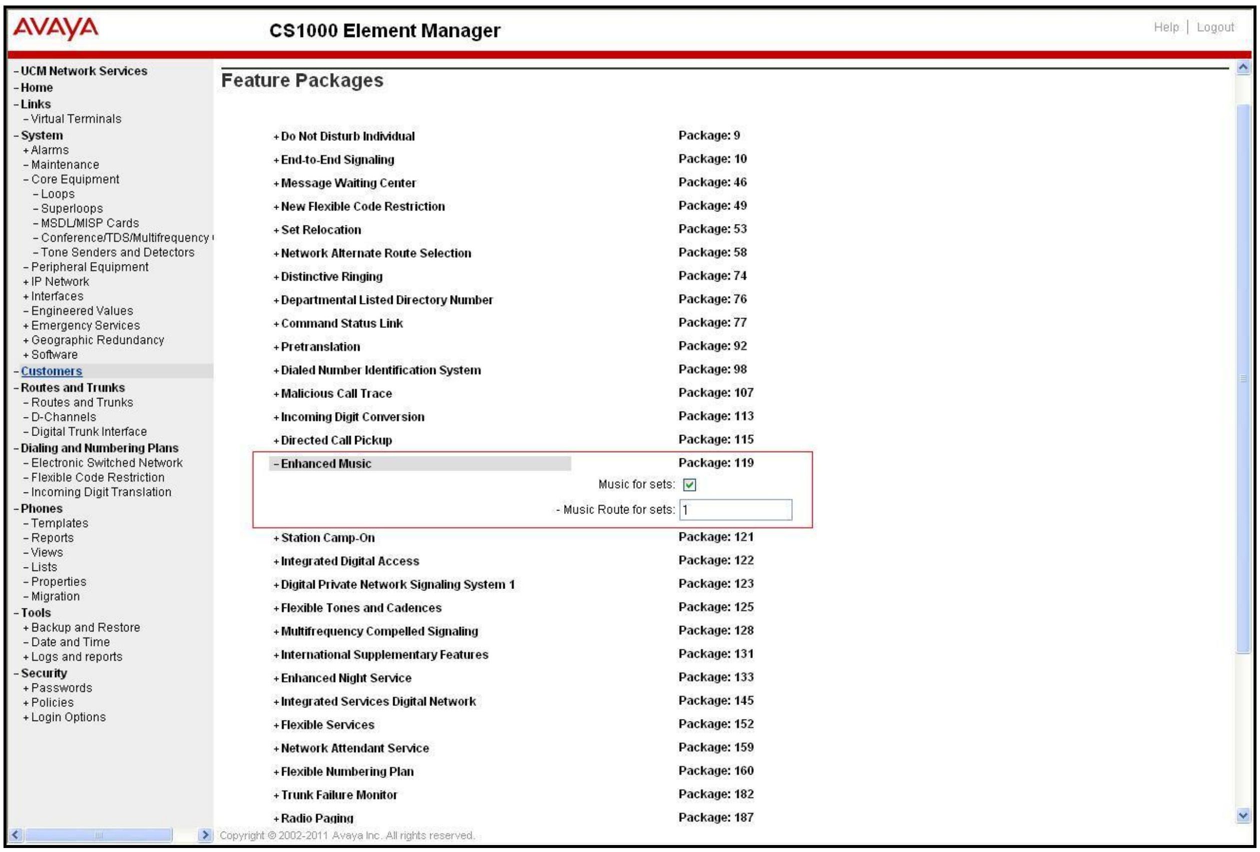


Figure 5.31 Enable Music for Customer 00

5.5.6. Administer Virtual SIP Routes

a) Select **Routes and Trunks** -> **Routes and Trunks** from the left pane to display the **Routes and Trunks** screen. In this example, **Customer 0** is being used. Click on the **Add route** button as shown in

Figure 5.32.

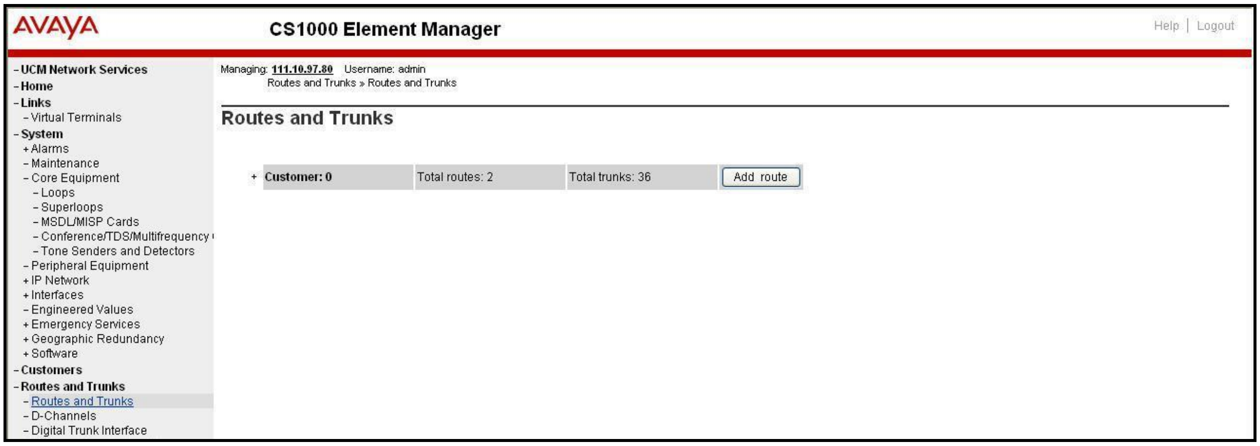


Figure 5.32 Add route

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

Figure 5.33.

- **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 four additional fields to appear.
- For the **Zone for codec selection and bandwidth management (ZONE)** field, enter 255 (created in **Section 5.4.2**).
- For the **Node ID of signaling server of this route (NODE)** field, enter the node number 1000 (created in **Section 5.2.1**).
- Select **SIP** (SIP) from the drop-down list for the **Protocol ID for the route (PCID)** field.
- Check the **Integrated Services Digital Network option (ISDN)** checkbox to enable additional fields to appear. Enter the following values for the specified fields, and retain the default values for the remaining fields. Scroll down to the bottom of the screen.
    - **Mode of operation (MODE)**: Route uses **ISDN Signalling Link (ISLD)**
    - **D channel number (DCH)**: D-Channel number 100 (created in **Section 5.5.3**)
    - **Network calling name allowed (NCNA)**: Check the field.
    - **Network call redirection (NCRD)**: Check the field.
    - **Insert ESN access code (INAC):** Check the field.

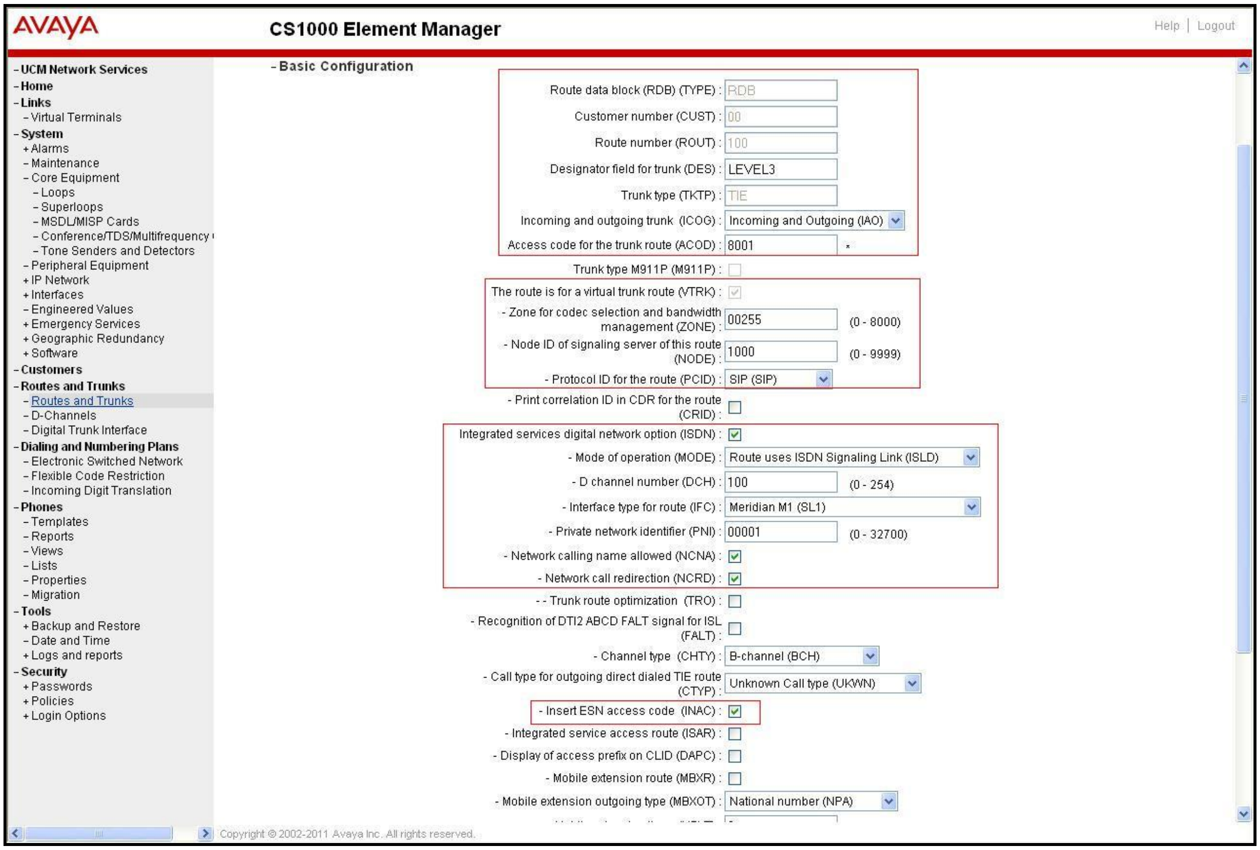


Figure 5.33 Route Configuration Details Pages 1

- Click on **Basic Route Options**, check the **North American toll scheme (NATL)** and **Incoming DID digit conversion on this route (IDC)**, input **DCNO 1** for both **Day IDC Tree Number** and **Night IDC Tree Number** as shown in
- **Figure** 5.34.

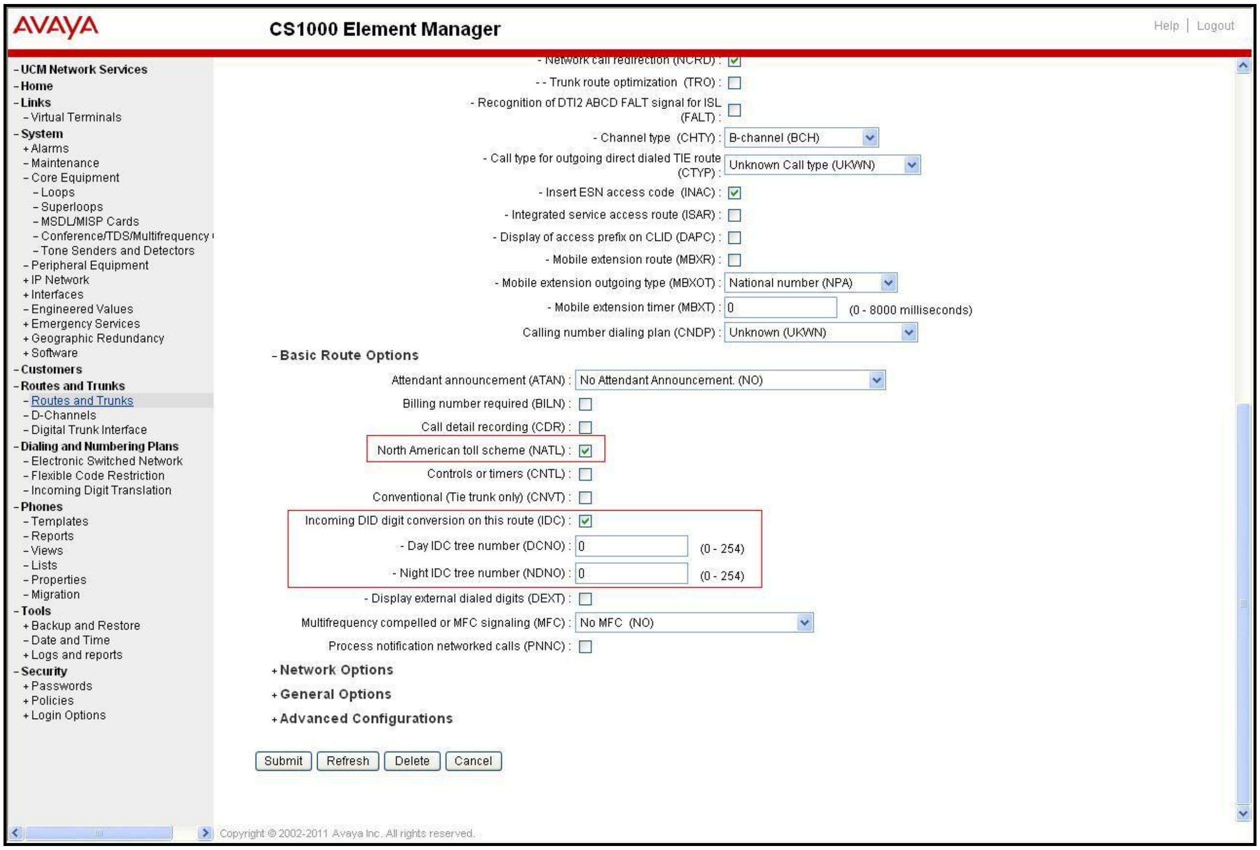


Figure 5.34 Route Configuration Details Pages 2

- Click on **Advance Configurations**; check **Music-on-hold** to enable music on hold on the route. Input music route 1 to the boxes as shown in
- **Figure** 5.35. The CS1000E system has been pre-configured with route 1 as a music route.

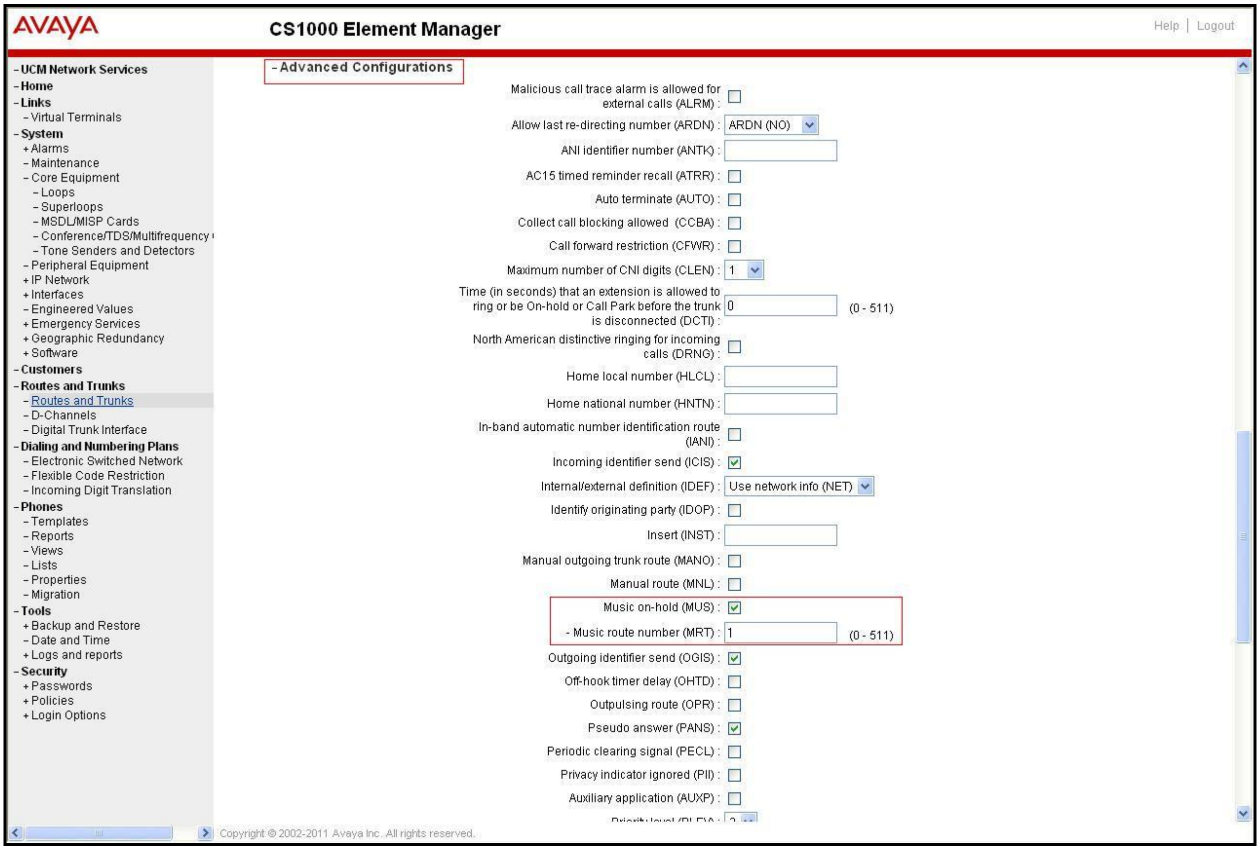


Figure 5.35 Route Configuration Details Pages 3

c) Click on the **Submit** button.

5.5.7. Administer Virtual Trunks

a) Continue **Section 5.5.6**, after click **Submit**, the **Routes and Trunks** screen is displayed and updated with the newly added route. In the example, Route 100 was being added. Click on the **Add trunk** button next to the newly added route 100 as shown in

Figure 5.36.

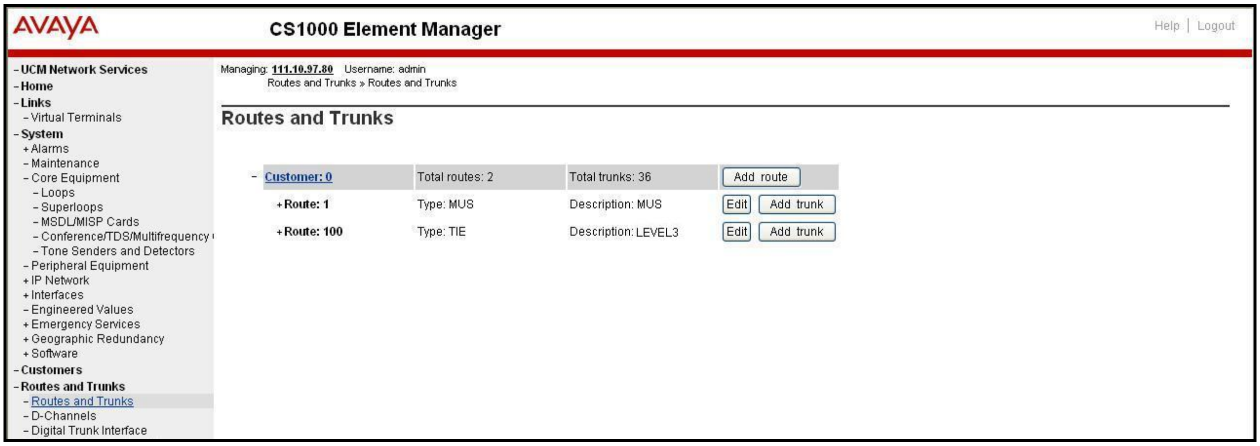


Figure 5.36 Route and Trunks

b) The **Customer 00, Route 100, Trunk 1 Property Configuration** screen is displayed in

Figure 5.37**.** Enter the following values for the specified fields and retain the default values for the remaining fields. 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 on the **Edit** button as shown in

Figure 5.37.

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

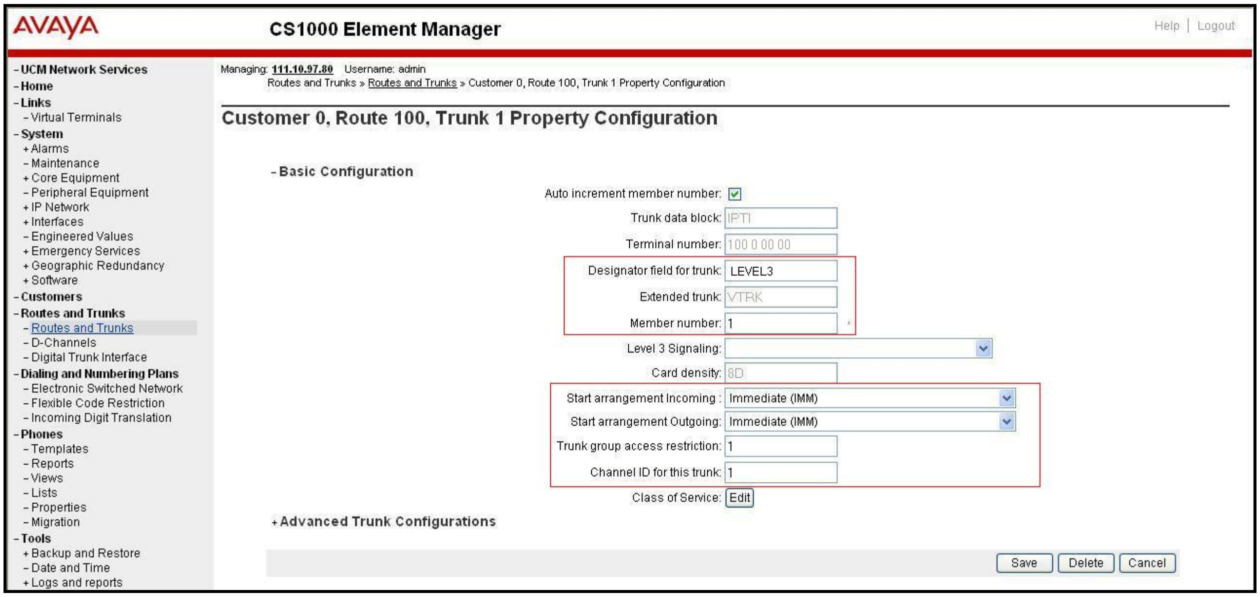


Figure 5.37 New Trunk Configuration Details

c) For **Media Security**, select **Media Security Never** (**MSNV)**. Enter the remaining values for the specified fields as shown in

Figure 5.38. Scroll down to the bottom of the screen and click **Return Class of Service** and then click on the **Save** button (not shown).

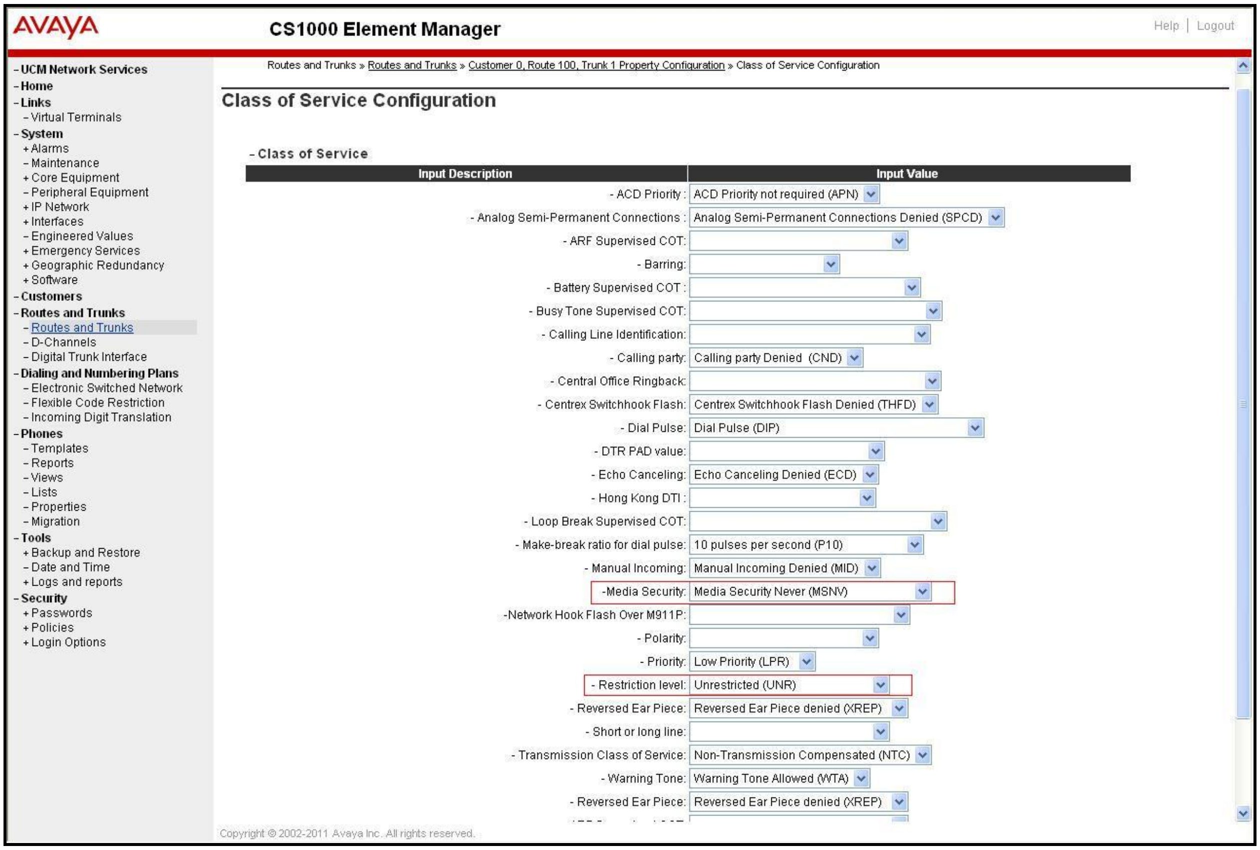


Figure 5.38 Class of Service Configuration Details Page

5.5.8. Administer Calling Line Identification Entries

a) Select Customers → 00 → ISDN and ESN Networking. Click on Calling Line Identification Entries as shown in
Figure 5.39.

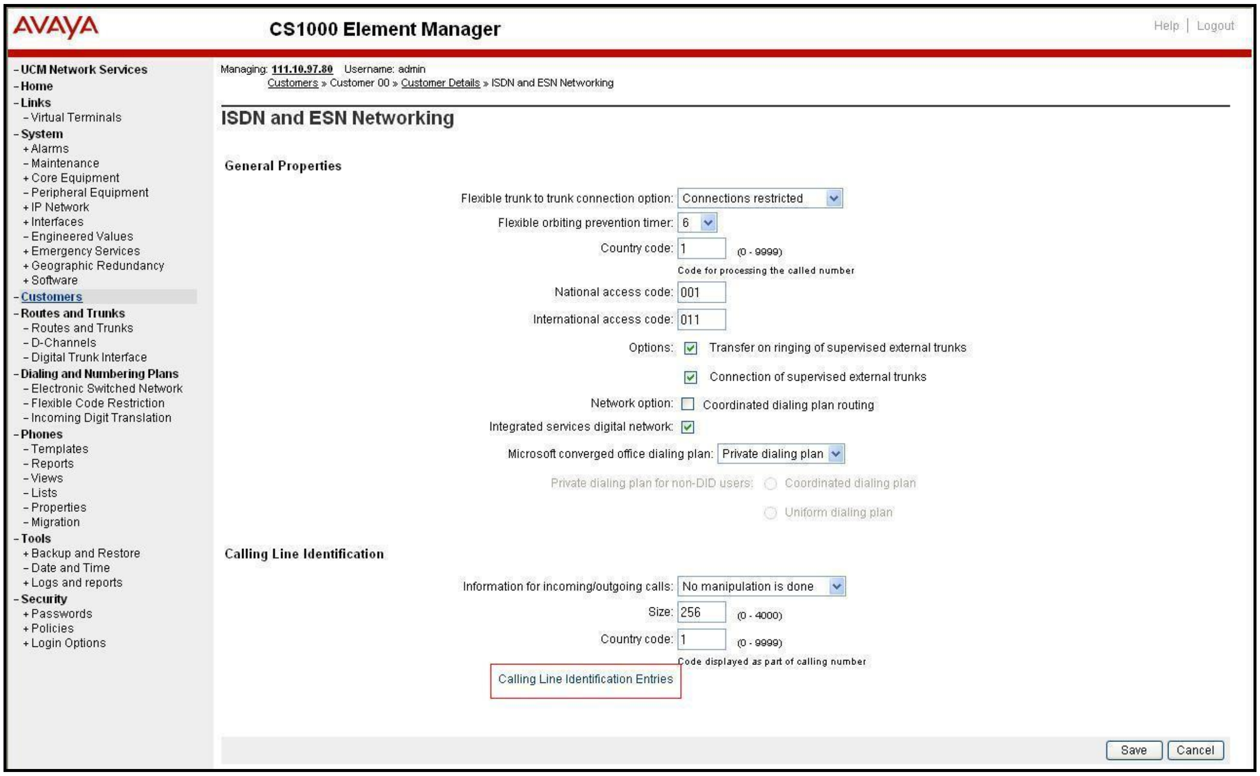


Figure 5.39 ISDN and ESN Networking

b) Click on **Add** as shown in
Figure 5.40.

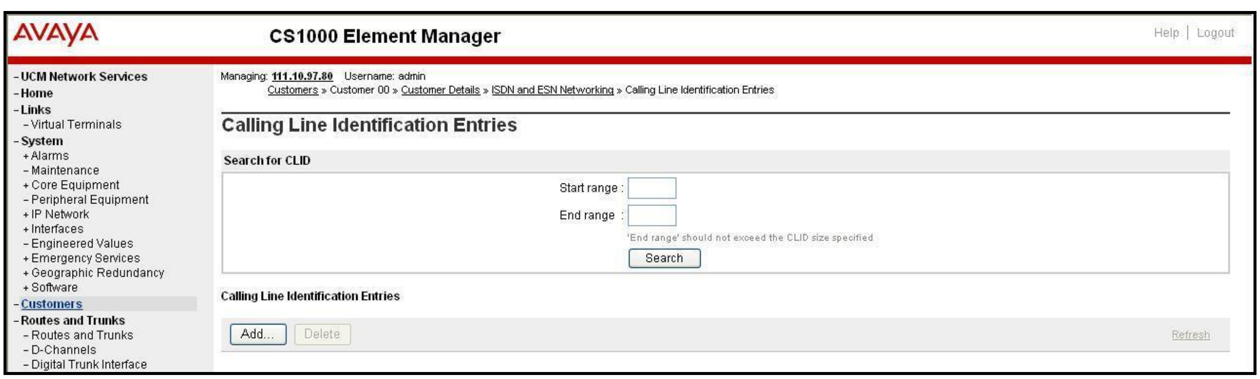


Figure 5.40 Calling Line Identification Page

c) Add entry **0** as shown in
Figure 5.41

- **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** will be 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** will be used for call display purpose for Call Type = National (NPA).
- **Local Steering Code**: input prefix digits assigned by Service Provider, in this case it is 6 digits - 716261. This **Local Steering Code** will be used for call display purpose for Call Type = Local Subscriber (NXX).
- **Calling Party Name Display**: Uncheck for **Roman characters**.

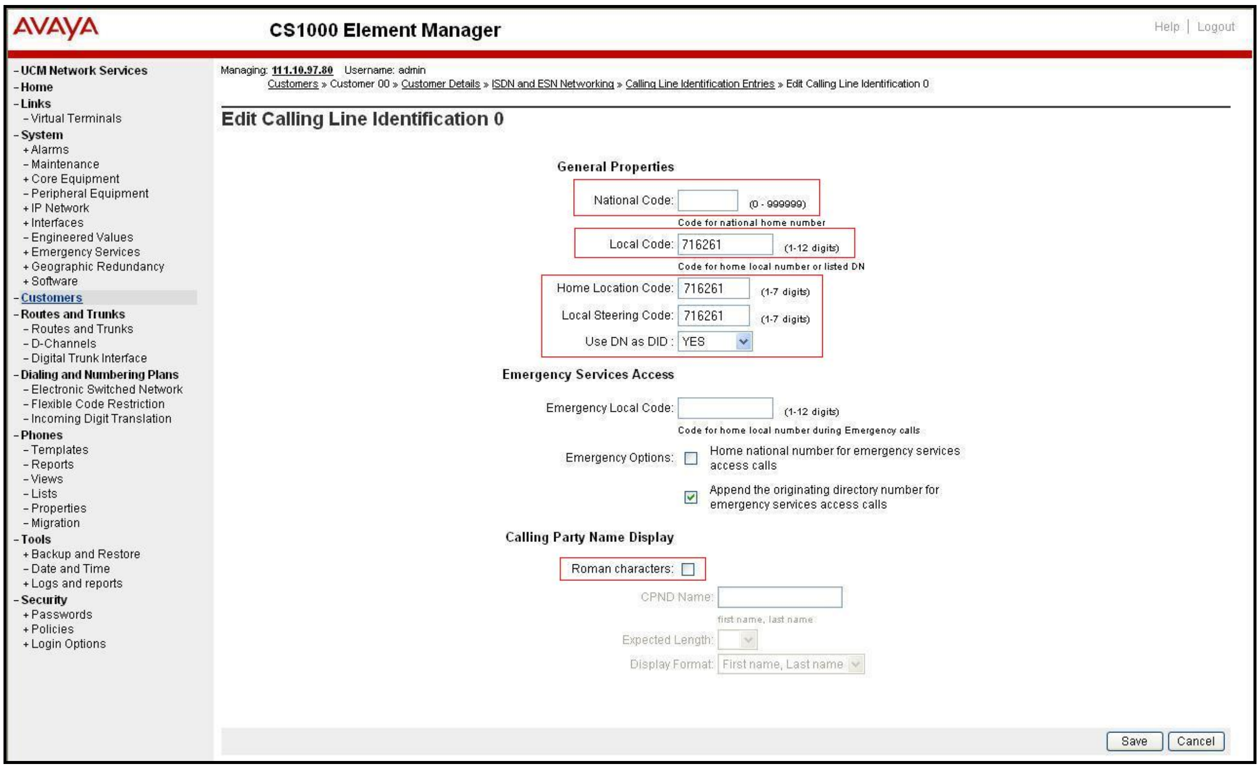


Figure 5.41 Edit Calling Line Identification 0

d) Click on **Save**.

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 trunk.
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 0
OPT
…
TRNX yes
EXTT yes
…
```

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. **Select ESN Access Code and Parameters (ESN)** as shown in
Figure 5.42.

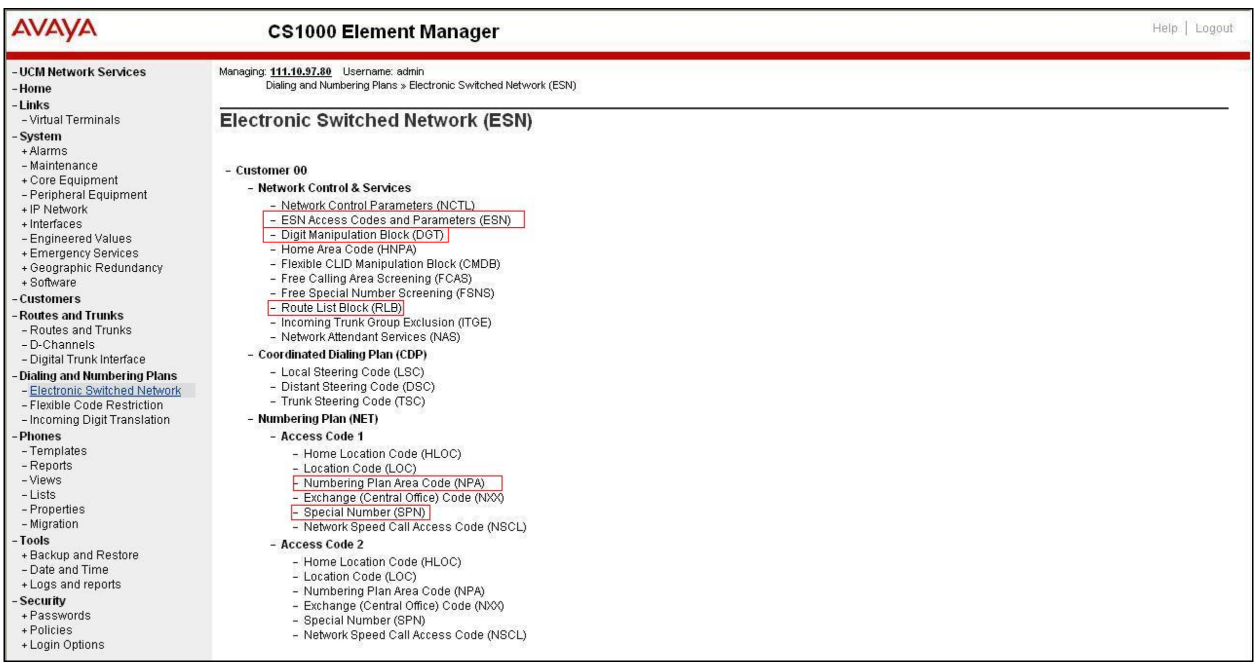


Figure 5.42 Electronic Switch Network (ESN)

b) In the **ESN Access Codes and Basic Parameters** page, define **NARS/ BARS Access Code 1** as shown in **Figure 5.43**.

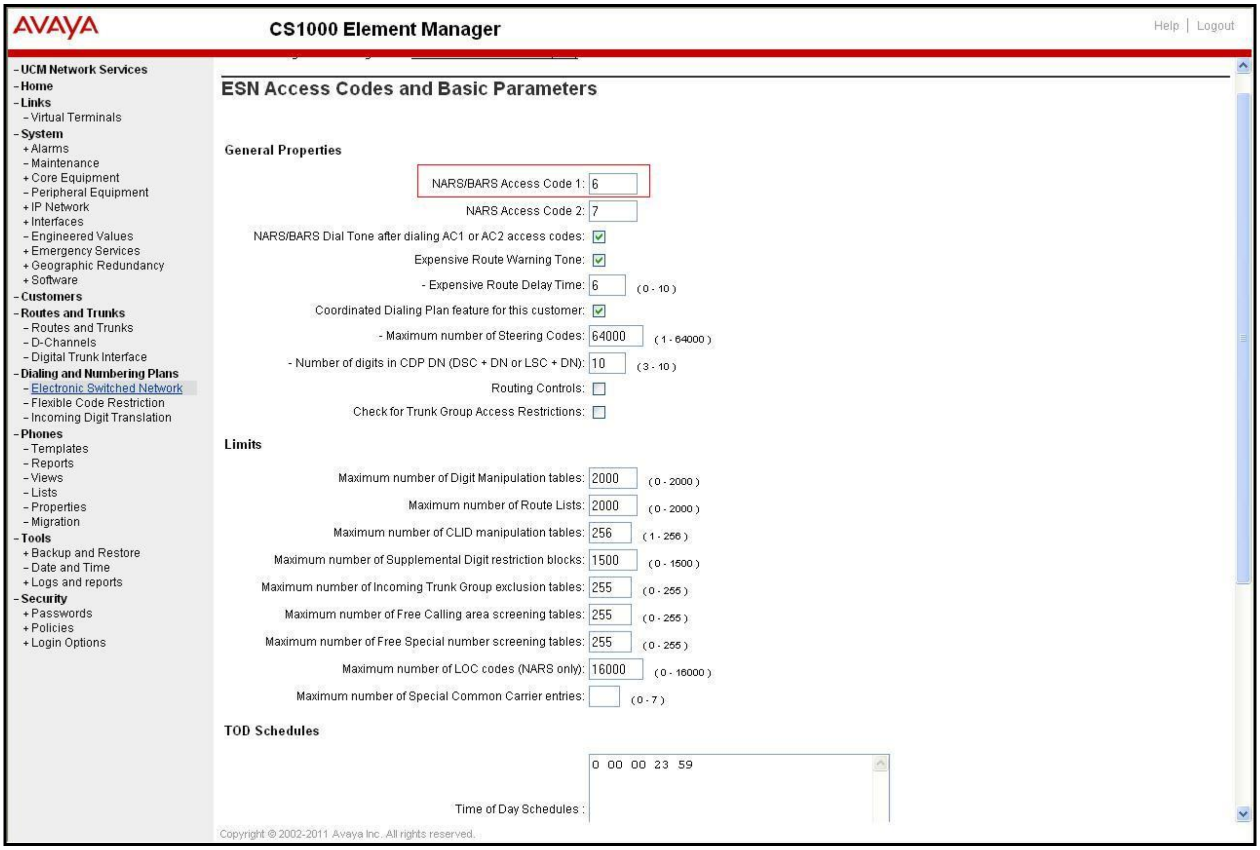


Figure 5.43 ESN Access Codes and Basic Parameters

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

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

a) Login Call Server CLI (please 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): 35717857    USED U P: 8241949 920063    TOT: 44879869
DISK SPACE NEEDED: 1697 KBYTES
REQ: chg
TYPE: net_data
CUST 0
OPT
AC2 xnpa xspn
FNP
```

```
CLID
ISDN
…
```

c) Verify Customer Net_Data block by using LD 21

```
>ld 21
PT1000

REQ: prt
TYPE: net
TYPE NET_DATA
CUST 0

TYPE NET_DATA
CUST 00
OPT RTA
AC1 INTL NPA SPN NXX LOC
AC2
FNP YES
…
```

5.6.3. Digit Manipulation Block (DMI)

a) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen. Select **Digit Manipulation Block** (DGT) as shown in
Figure 5.42.
b) In the Choose a Digit Manipulation Block Index (DMI) Number field, select an available DMI from the drop-down list and click **to Add** as shown in
Figure 5.44.

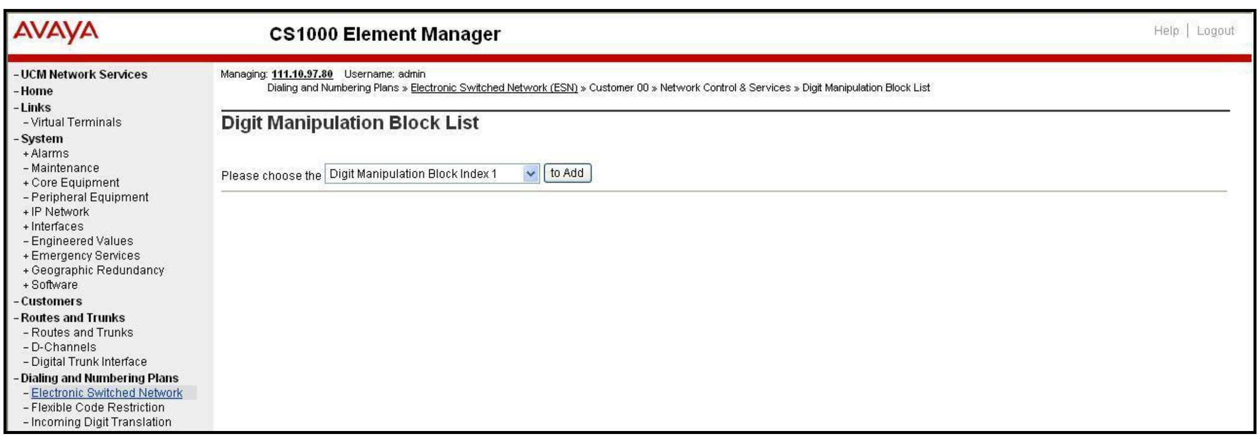


Figure 5.44 Digit Manipulation Block List

c) Enter **0** for the **Number of leading digits to be Deleted** (Del) field and select **NPA (NPA)** for the **Call Type to be used by the manipulated digits** (CTYP) and then click **Submit** as shown in

Figure 5.45.

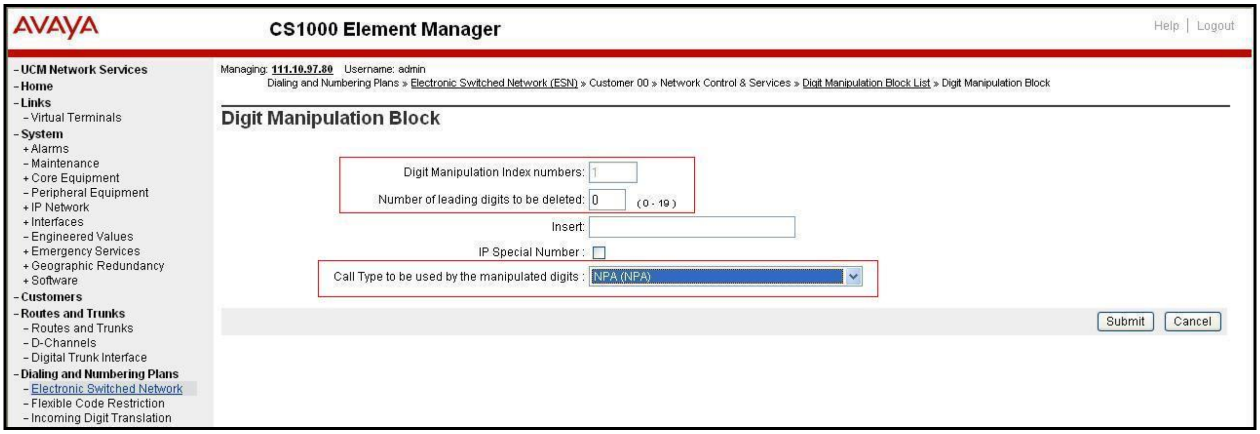


Figure 5.45 Digit Manipulation Block

5.6.4. Route List Block (RLB) (RLB 100)

This section shows how to add a RLB associated with the DMI created in **Section 5.6.3**.

a) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen. Select **Route List Block** (RLB) as shown in
Figure 5.42.

b) Select an available value in the textbox for the **route list index** and click on the "**to Add"** button (in this case is 100) as shown in
Figure 5.46.

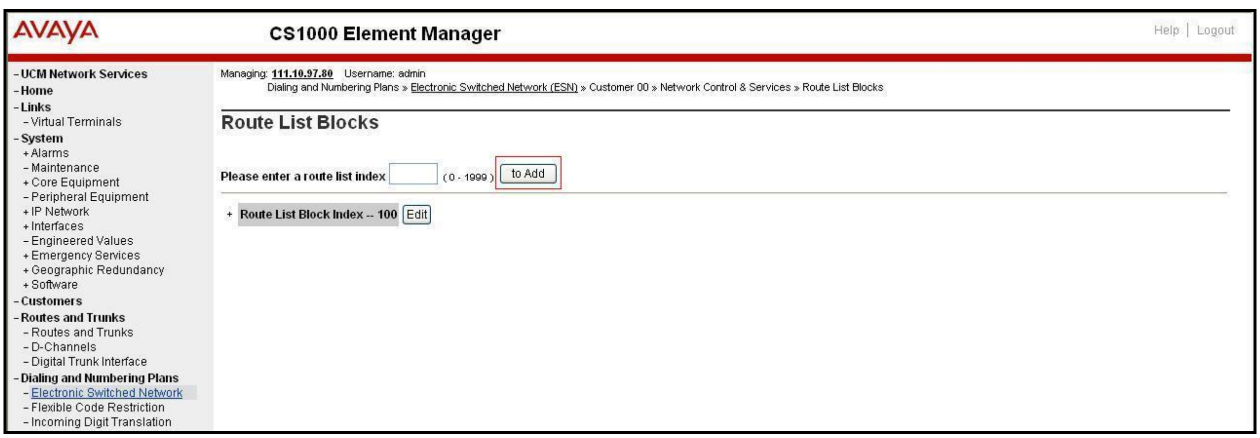


Figure 5.46 Route List Blocks

c) Enter the following values for the specified fields, and retain the default values for the remaining fields as shown in
Figure 5.47. Scroll down to the bottom of the screen, and click on the **Submit** button.

- **Route number** (ROUT): 100 (created in **Section 5.5.6**)

- **Digit Manipulation Index** (DMI): 0 (created in **Section 5.6.3**)

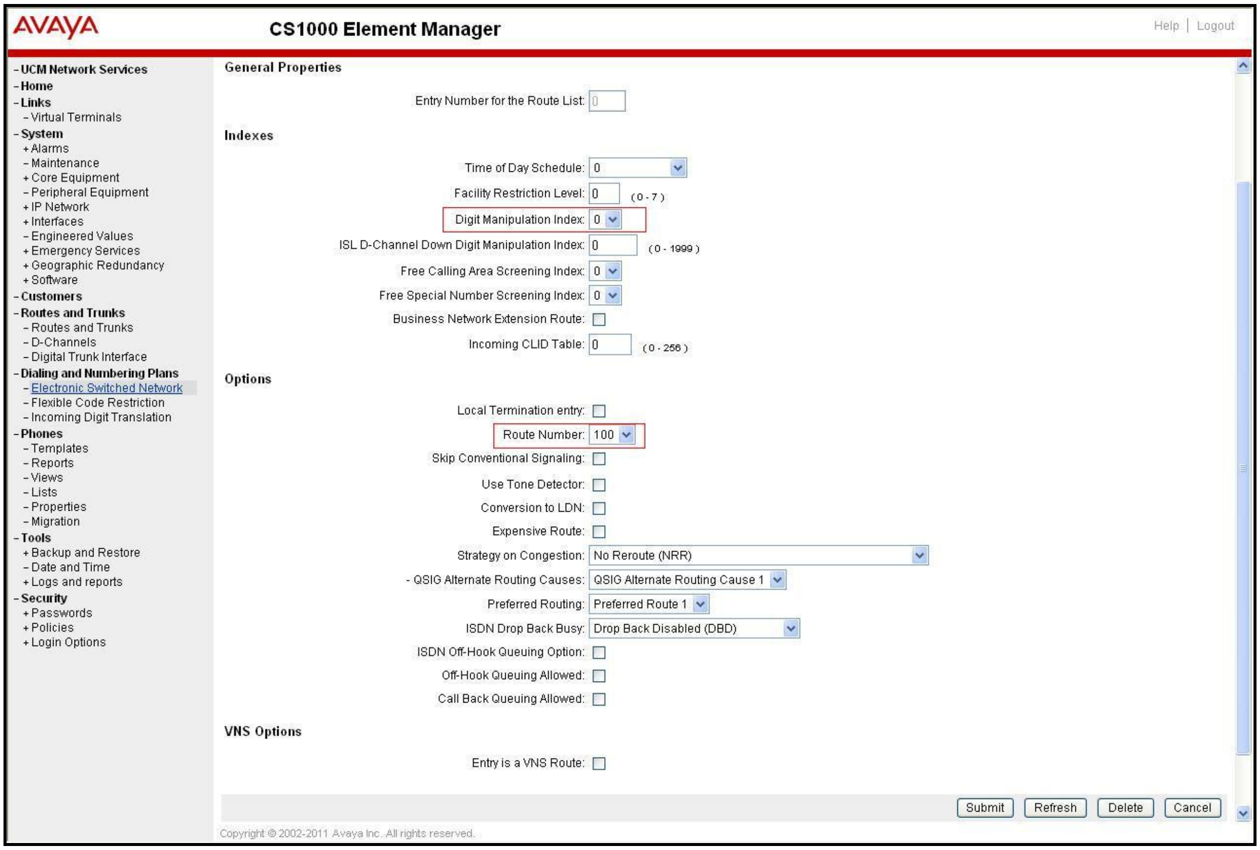


Figure 5.47 Route List Blocks Configuration Details

5.6.5. Inbound Call Digit Translation

This section describes the steps for receiving the calls from PSTN via the Level 3 system.

a) Select **Dialing and Numbering Plans** -> **Incoming Digit Translation** from the left pane to display the **Incoming Digit Translation** screen. Click on the **Edit IDC** button as shown in **Figure** 5.48

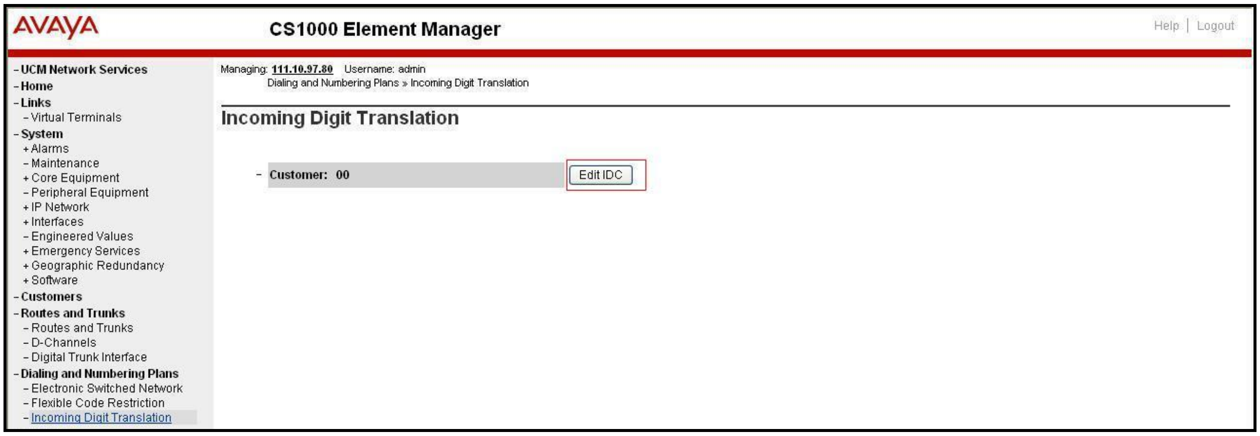


Figure 5.48 Incoming Digit Translation

b) Click on the **New DCNO** to create the digit translation mechanism. In this example, Digit Conversion Tree Number (**DCN0**) 1 has been created as shown in
Figure 5.49.

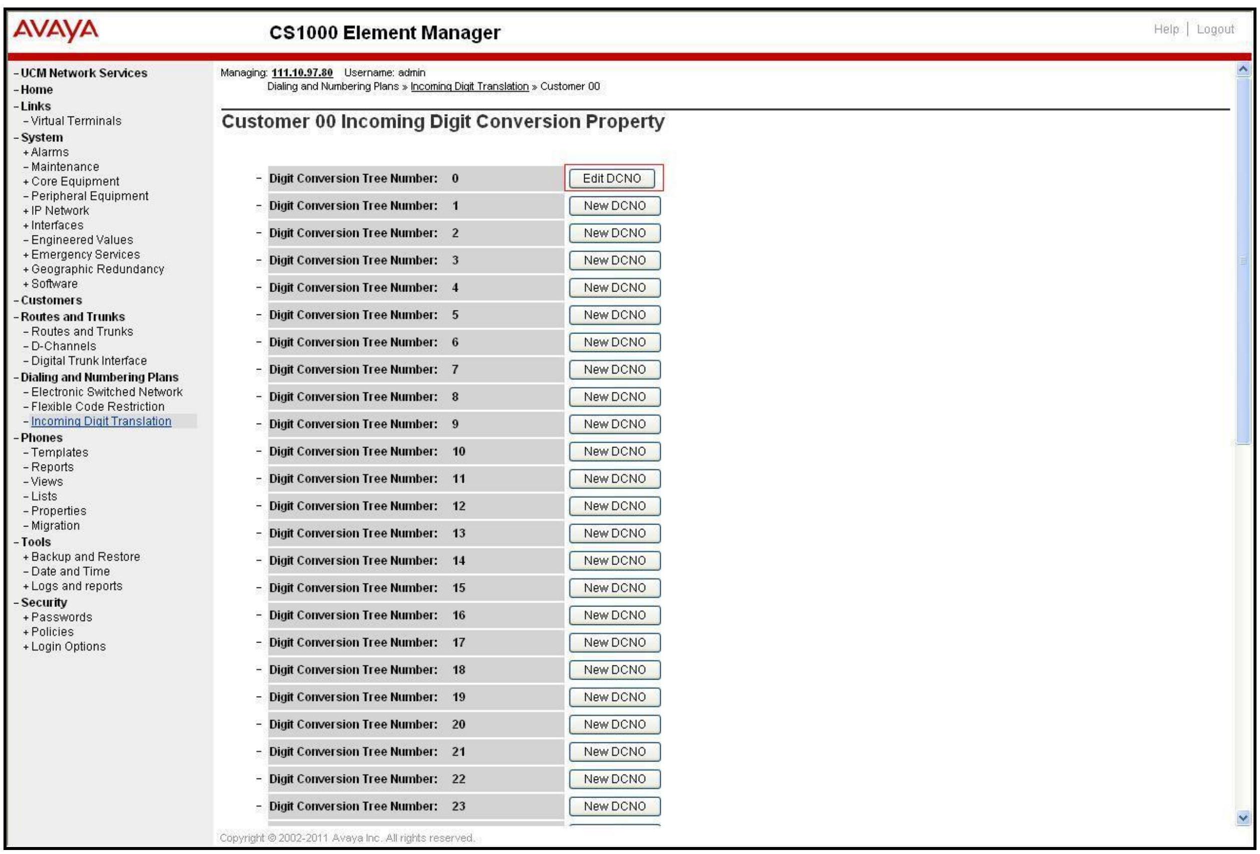


Figure 5.49 Incoming Digit Conversion Property

c) Detail configuration of the **DCNO** is shown in

Figure 5.50. The **Incoming Digits** can be added to map to the **Converted Digits** which would be the CS1000E system phones DN. This **DCN0** has been assigned to route 100 as shown in **Figure** 5.34.
In the following configuration, the incoming call from PSTN with the prefix 71626112xx will be translated to CS1000E DN 12xx. The DID 7162611224 is translated to 1700 for Voicemail accessing purpose.

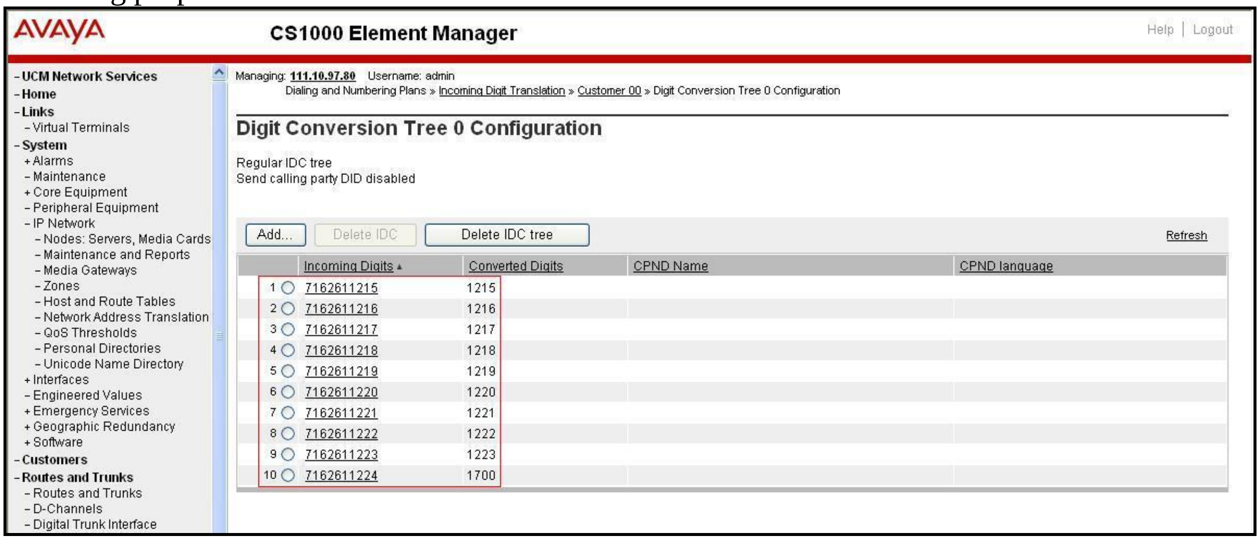


Figure 5.50 Digit Conversion Tree Configuration

5.6.6. Outbound Call - Special Number Configuration.

There are special numbers which have been configured to be used for this testing such as; **0** to reach Service Provider operator, **0+10** digits to reach Service Provider operator assistant, **011** prefix for international call, **1** for national long distance call, **411**, **911** and so on.

a) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen. Select **Special Number** (SPN) as shown in **Figure 34**.
b) Enter SPN and then click on the "**to Add**" button.
Figure 5.51 shows all the special numbers were used for this testing.

Special Number: 0

- **Flexible length:** 0 (flexible, unlimited and accept the character # to ending dial number)
- **CallType:** NONE
- **Route list index:** 100, created in **Section 5.6.4**

Special Number: 1

- **Flexible length:** 0 (flexible, unlimited and accept the character # to ending dial number)
- **CallType**: NATL
- **Route list index**: 100, created in **Section 5.6.4**

Special Number: 411

- **Flexible length**: 3
- **CallType**: NATL
- **Route list index**: 100, created in **Section 5.6.4**

Special Number: 911

- **Flexible length**: 3
- **CallType**: NATL
- **Route list index**: 100, created in **Section 5.6.4**

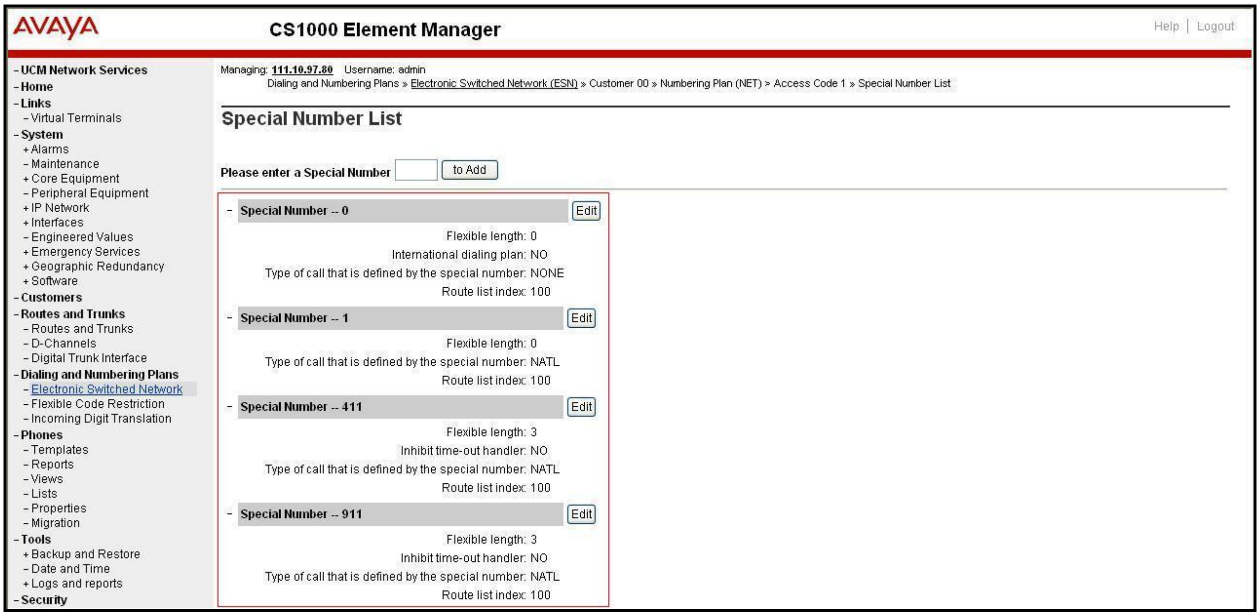


Figure 5.51 Special Number List

5.6.7. Outbound Call - Numbering Plan Area (NPA)

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

a) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen. Select **Numbering Plan Area Code** (NPA) as shown in

Figure 5.42.

b) Enter area code desired in the textbox and click on the "**to Add**" button.

Figure 5.52 shows NPA numbers **613** configured for this testing. These codes are associated to SIP route 100.

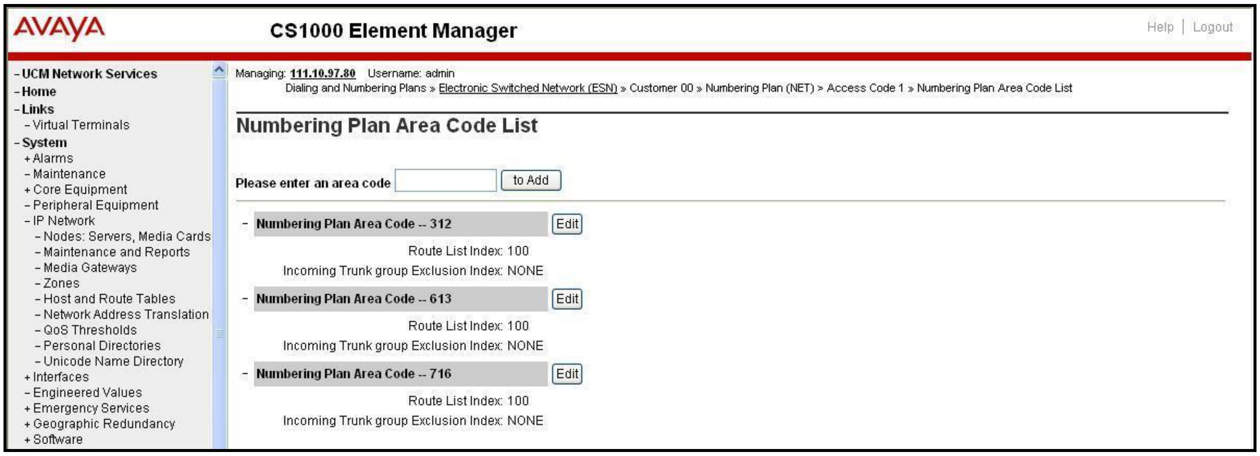


Figure 5.52 Numbering Plan Area Code List

5.7. Administer Phone

This section describes the creation of CS1000E clients used in this testing configuration.

5.7.1. Phone creation

a) Refer to **Section 5.5.4** to create a virtual super-loop - **108** used for IP phone.
b) Refer to **Section 5.4.1** to create a bandwidth zone - **10** for IP phone.
c) Login Call Server CLI (please refer to **Section 5.1.2** for more detail).
d) Create an IP phone by using LD 11.

```
REQ: prt
TYPE: 2004p1

TN   96 0 0 1
DATE
PAGE
DES
MODEL_NAME
EMULATED
KEM_RANGE

DES  PHONE
TN   096 0 00 01  VIRTUAL
TYPE 2004P1
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 00010
CUR_ZONE 00010
MRT
ERL  0
ECL  0
FDN  16139675204
```

```
TGAR 0
LDN  NO
NCOS 0
SGRP 0
RNPG 0
SCI  0
SSU
LNRS 16
XLST
SFLT NO
CAC_MFC 0
CLS  UNR FBA WTA LPR MTD FNA HTA ADD HFD CRPD
     MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LNA CNDA
     CFTD SFA MRD DDV CNID CDCA MSID DAPA BFED RCBD
     ICDD CDMD LLCN MCTD CLBD AUTU
     GPUD DPUD DNDD CFXA ARHD CLTD ASCD
     CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
     UDI RCC HBTD AHD IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
     DRDD EXR0
     USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
     FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD
     KEM2 MSNV FRA  PKCH MUTA MWTD DVLD CROD ELCD
CPND_LANG ENG
RCO  0
HUNT 16139675204
LHK  0
PLEV 02
PUID
UPWD
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY  00 SCR 1216 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME Level3 i2004P1
            XPLN 13
            DISPLAY_FMT FIRST,LAST
     01 MSB
     02
     03
     04
     05
     06
     07
     08
     09
     10
```

```
11
12
13
14
15
16
17 TRN
18 AO6
19 CFW 16  616139675204
20 RGA
21 PRK
22 RNP
23
24 PRS
25 CHG
26 CPN
27
28
29
30
31
```

5.7.2. Enable Privacy for Phone

In this section, it shows how to enable Privacy for a phone by changing its class of service (CLS). By modifying the configuration of the phone created in **Section 5.7.1**, the display of the outbound call will be changed appropriately. The privacy for a single call can be done by configuring per-call blocking and a corresponding dialing sequence, for example *67. The resulting SIP privacy setting will be the same in either case.

a) To hide display name, set CLS to **namd**. CS1000E will include “Privacy:user” in SIP message header before sending to Service Provider.

```
>ld 11
REQ: chg
TYPE: 2004p1
TN   96 0 0 1
ECHG yes
ITEM cls namd
ITEM
…
```

b) To hide display number, set CLS to **ddgd**. CS1000E will include “Privacy:id” in SIP message header before sending to Service Provider.

```
>ld 11
REQ: chg
TYPE: 2004p1
TN   96 0 0 1
ECHG yes
ITEM cls ddgd
…
```

c) To hide display name and number, set CLS to **namd, ddgd**. CS1000E will include “Privacy:id, user” in SIP message header before sending to Service Provider.

```
>ld 11
REQ: chg
TYPE: 2004p1
TN   96 0 0 1
ECHG yes
ITEM cls namd ddgd
…
```

d) To allow display name and number, set CLS to **nama, ddga**. CS1000E will send header “Privacy:none” to Service Provider.

```
>ld 11
REQ: chg
TYPE: 2004p1
TN   96 0 0 1
ECHG yes
ITEM cls nama ddga
…
```

5.7.3. Enable Call Forward for Phone

In this section, it shows how to configure Call Forward feature at the system level and phone level.

a) Select **Customer → 00→ Call Redirection**. The Call Redirection page is shown as **Figure** 5.53.

- **Total redirection count limit**: **0** (unlimited)
- **Call Forward: Originating**
- **Number of normal ring cycle of CFNA: 4**

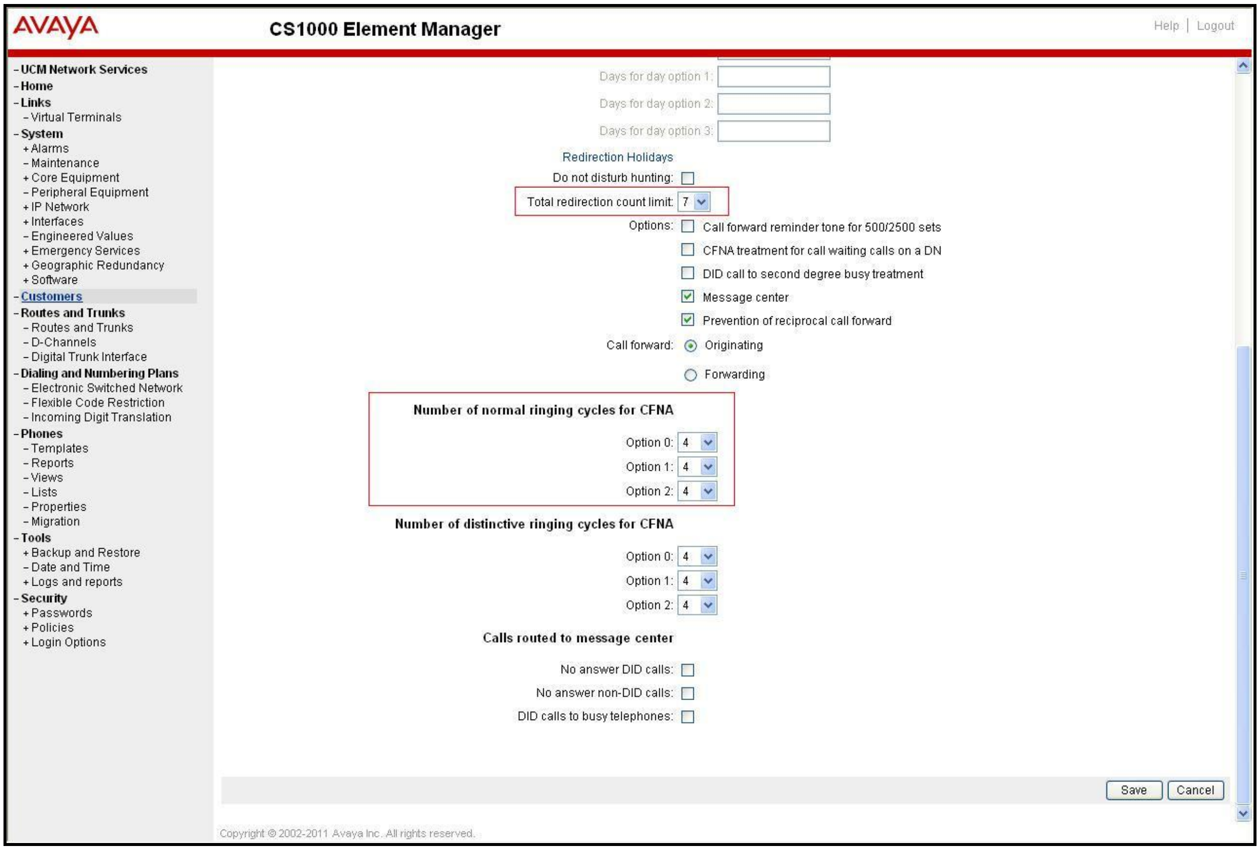


Figure 5.53 Call Redirection

b) To enable **Call Forward All Call** (**CFAC**) for phone over trunk by using LD 11, change its CLS to **CXFA** then program the forward number on the phone set. Following is the configuration of a phone that has CFAC enabled with forwarding number as 616139675204.

```
REQ: prt
TYPE: 2004p1
TN   96 0 0 1
DATE
PAGE
DES
MODEL_NAME
```

```
EMULATED

DES  PHONE
TN   96 0 00 01  VIRTUAL
TYPE 2004P1
…
CLS  UNR FBA WTA LPR MTD FNA HTA TDD HFD CRPD
     MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LNA CNDA
     CFTD SFA MRD DDV CNID CDCA MSID DAPA BFED RCBD
     ICDA CDMA LLCN MCTD CLBD AUTU
     GPUD DPUD DNDD CFXA ARHD CLTD ASCD
…
     19 CFW 16  616139675204
…
```

c) To enable **Call Forward Busy (CFB)** for phone over trunk by using LD 11, change its CLS to **FBA, HTA** then program the forward number as **HUNT**. Following is the configuration of a phone that has CFB enabled with forward number as 616139675204.

```
REQ: prt
TYPE: 2004p1
TN   96 0 0 1
DATE
PAGE
DES
MODEL_NAME
EMULATED

DES  PHONE
TN   96 0 00 01  VIRTUAL
TYPE 2004P1
…
CLS  UNR FBA WTA LPR MTD FNA HTA TDD HFD CRPD
     MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LNA CNDA
     CFTD SFA MRD DDV CNID CDCA MSID DAPA BFED RCBD
…
HUNT 616139675204
…
```

d) To enable **Call Forward No Answer (CFNA)** for phone over trunk by using LD 11, change its CLS to **FNA, SFA** then program the forward number as **FDN**. Following is the configuration of a phone that has CFNA enabled with forward number as 616139675204.

```
REQ: prt
TYPE: 2004p1
TN   96 0 0 1
DATE
PAGE
```

```
DES
MODEL_NAME
EMULATED

DES  PHONE
TN   96 0 00 01  VIRTUAL
TYPE 2004P1
…
FDN  616139675204
…
CLS  UNR FBA WTA LPR MTD FNA HTA TDD HFD CRPD
     MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LNA CNDA
     CFTD SFA MRD DDV CNID CDCA MSID DAPA BFED RCBD
   …
```

5.7.4. Enable Call Waiting for Phone

In this section, it shows how to configure Call Waiting feature at phone level.

a) Login Call Server CLI (please refer to **Section 5.1.2** for more detail).
b) Configure Call Waiting feature for phone by using LD 11 to change CLS to **HTD**, **SWA** and adding a **CWT** key.

```
REQ: prt
TYPE: 2004p1
TN   96 0 0 1
DATE
PAGE
DES
MODEL_NAME
EMULATED
KEM_RANGE

DES  2004P1
TN   96 0 00 00  VIRTUAL
TYPE 2004P1
…
CLS  UNR FBD WTA LPR MTD FNA HTD TDD HFD CRPD
     MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWA LNA CNDA
…
KEY  00 SCR 5904 0     MARP
       CPND
         CPND_LANG ROMAN
           NAME Level3 i2004P1
           XPLN 13
           DISPLAY_FMT FIRST,LAST
     01 CWT
…
```

6. Administer Avaya Aura® Session Manager

In this section, it shows how to configure the routing on Session Manager. It is assumed that the Session Manager has been successfully deployed and connected to System Manager. The System Manager is the web interface to configure the Session Manager.

6.1. Create a SIP domain name

This section shows how to create a new SIP domain name for this test configuration. The Session Manager uses this domain name to route the call from Level 3 to CS1000E and vice versa.

a) Login to System Manager. Open the web browser then login with user "admin" and appropriate password as show in

Figure 6.1.

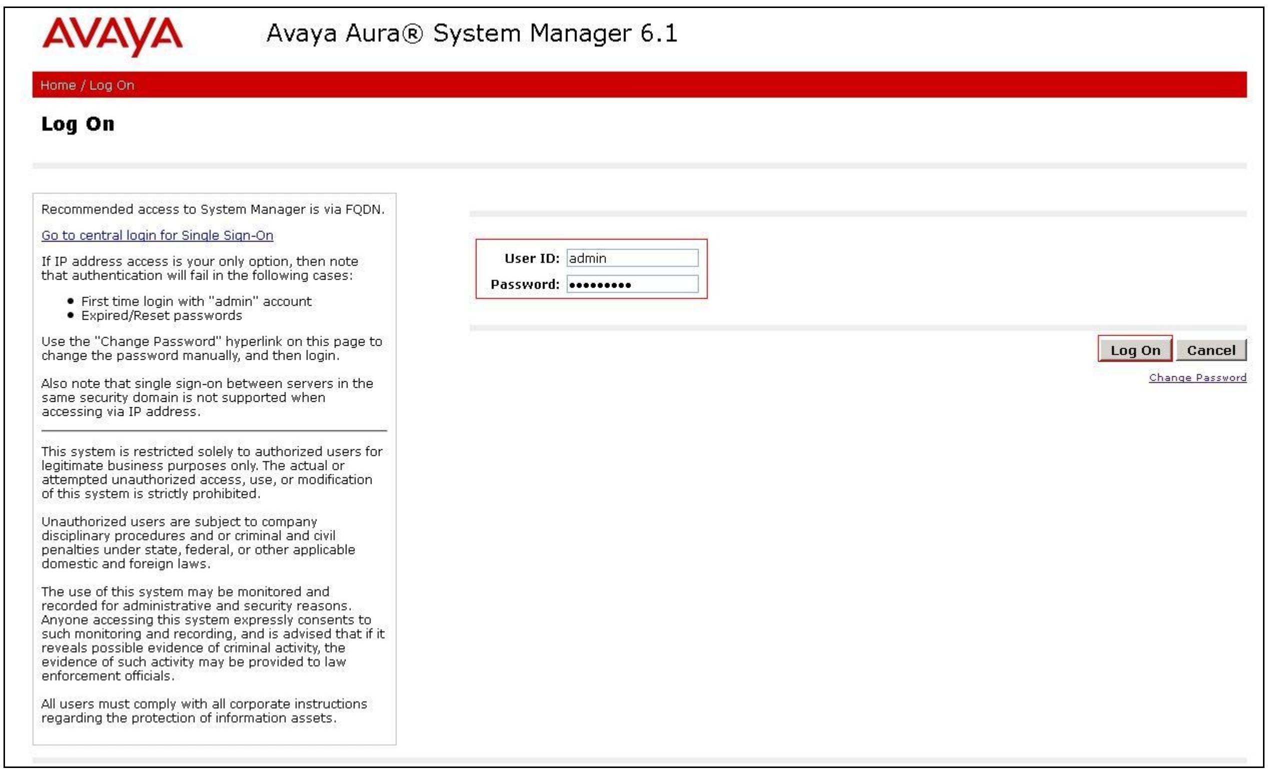


Figure 6.1 Login to System Manager

b) The System Manager home page displays as

Figure 6.2. Select **Routing** to configure the **Network Routing Policy**.

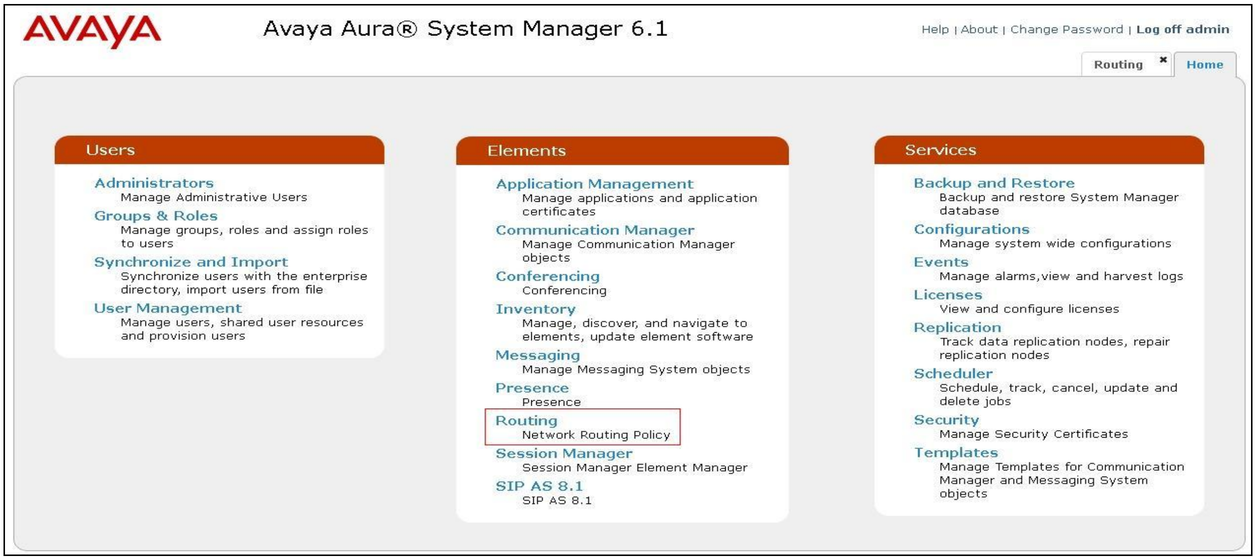


Figure 6.2 Select Routing to configure Network Routing Policy

c) In the **Introduction to Network Routing Policy** page (not shown), click **Domains** link on the left menu to open **Domains - Domain Management** page. Then click button **New** (not shown) to add a new test domain.

Figure 6.3 shows domain **level-3.voip.com** was successfully added.

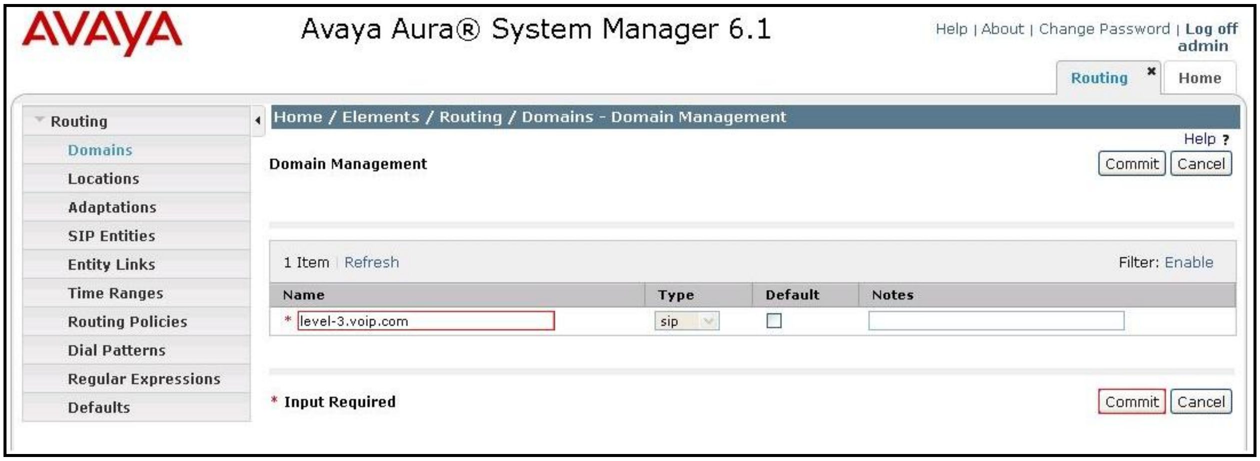


Figure 6.3 Adding SIP domain level-3.voip.com

d) Click **Commit**.

6.2. Create a Location

Other than domain name, Session Manager binds a SIP Entity to a Location for bandwidth and location management purposes. It inserts SIP header "P-Location" to tell the Service Provider where the call is made from.

The procedure to configure a location is as follows.

a) In the **Introduction to Network Routing Policy** page (not shown), click **Locations** link on the left menu to open **Locations - Location** page. Then click button **New** (not shown) to add a new test location.
Figure 6.4 shows location **Belleville,Ont,Ca** was successfully added with default settings in the red boxes.

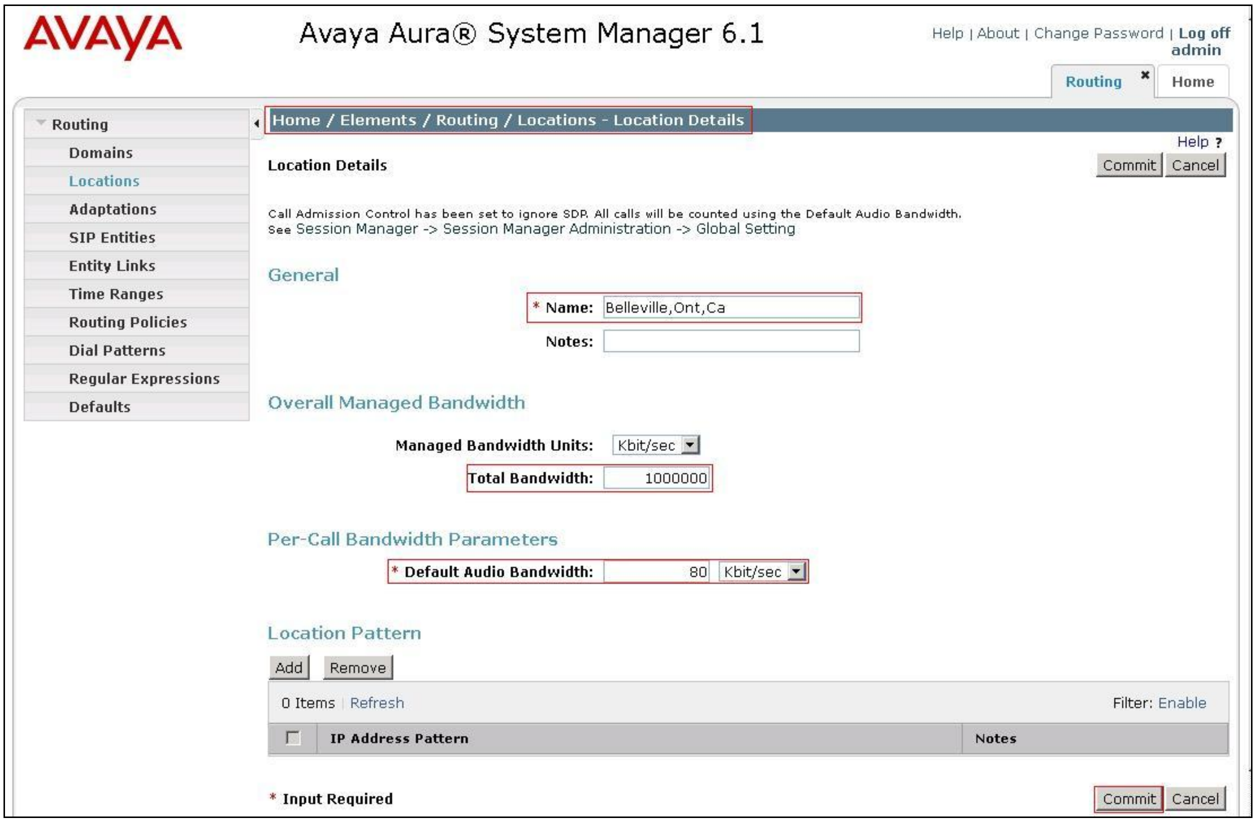


Figure 6.4 Adding a Location

b) Click **Commit**.

6.3. Create SIP Entity for Session Manager

This section shows how to configure System Manager to add a **SIP Entity** for Session Manager as a static gateway.

a) In the **Introduction to Network Routing Policy** page (not shown), click **SIP Entities** link on the left menu to open **SIP Entities – SIP Entities** page. Then click button **New** (not shown) to add a new entity for Session Manager.
Figure 6.5 shows entity **DevASM** was successfully added. The Session Manager was configured to use transport protocol UDP with port 5060.

- **Name**: DevASM
- **FQDN or IP Address**: 111.10.97.198
- **Type**: Session Manager
- **Location**: Belleville,Ont,Ca
- **Port**: 5060, **Protocol**: UDP
- **SIP Link Monitoring**: Use Session Manager Configuration

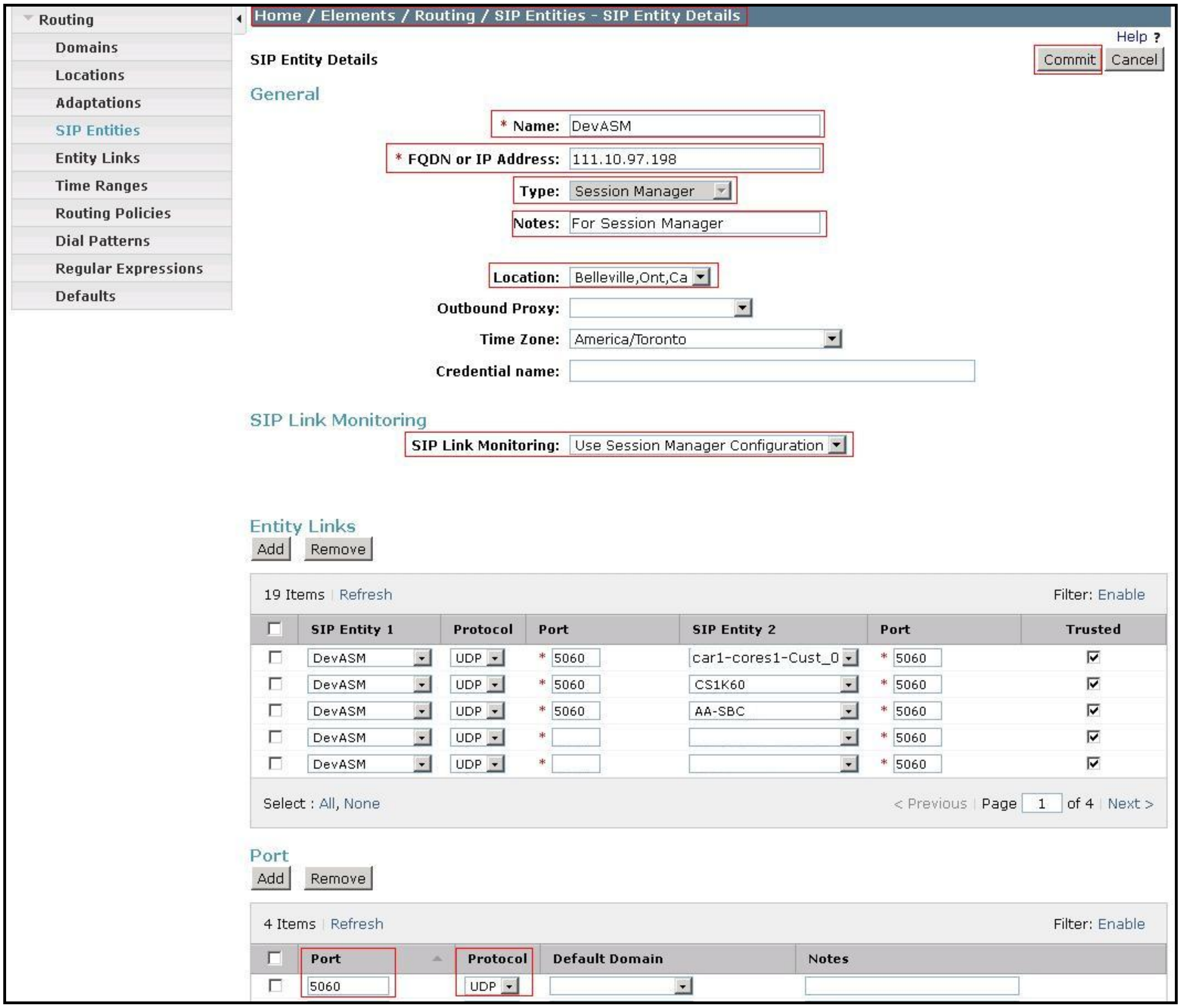


Figure 6.5 Adding SIP Entity for Session Manager

b) Click **Commit**.

Note: The IP Address used for SIP Entity - Session Manager has to be different than the IP address used for management interface of Session Manager. The management IP was associated to physical interface eth0 and was defined during software installation. While the IP for SIP Entity was associated to physical interface eth2.

6.4. Create SIP Entity for CS1000E SIP Gateway

This section shows how to configure System Manager to add a SIP Entity for CS1000E SIP Gateway.

a) In the **Introduction to Network Routing Policy** page (not shown), click **SIP Entities** link on the left menu to open **SIP Entities – SIP Entities** page. Then click button **New** (not shown) to add a new entity for CS1000E SIP Gateway.

The **Entity Links** configuration is to define the network connection between Session Manager and CS1000E SIP Gateway. In this testing, the trusted link was configured with protocol UDP and port 5060. The
Figure 6.6 shows SIP Entity **car1-cores1-Cust_0** was successfully added.

- **Name**: car1-cores1-Cust_0
- **FQDN or IP Address**: 111.10.97.154
- **Type**: Other
- **Adaptation:** CS1K Adaptation (configuration is shown in **Section 6.10**)
- **Location**: Belleville,Ont,Ca
- **SIP Link Monitoring**: Use Session Manager Configuration

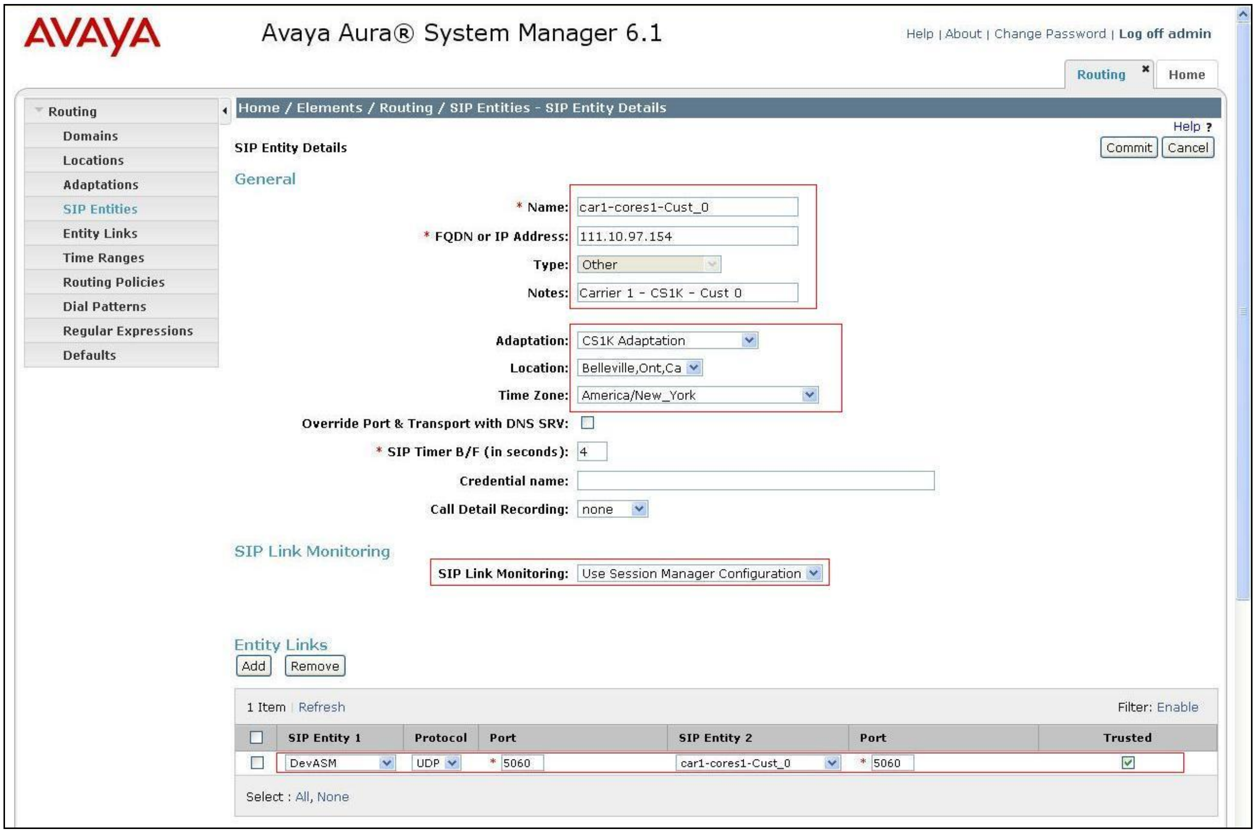


Figure 6.6 Adding SIP Entity for CS1000E SIP Gateway

b) Click **Commit**.

Note: In the **Entity Links** configuration, the option “**Trusted**” is mandatory.

6.5. Create SIP Entity for Avaya Aura® Packet SBC

This section shows how to configure System Manager to add a SIP Entity for Avaya Aura® SBC.

a) In the **Introduction to Network Routing Policy** page (not shown), click **SIP Entities** link on the left menu to open **SIP Entities – SIP Entities** page. Then click button **New** (not shown) to add a new entity for Avaya Aura® SBC.

The **Entity Links** configuration is to define the network connection between Session Manager and Avaya Aura® SBC. In this testing, the trusted link was configured with protocol UDP and port 5060. The
Figure 6.7 shows SIP Entity **AA-SBC** was successfully added.

- **Name**: AA-SBC
- **FQDN or IP Address**: 111.10.97.206
- **Type**: Other
- **Adaptation:** Diversion for Level 3 (configuration is shown in **Section 6.10**)
- **Location**: Belleville,Ont,Ca
- **SIP Link Monitoring**: Use Session Manager Configuration

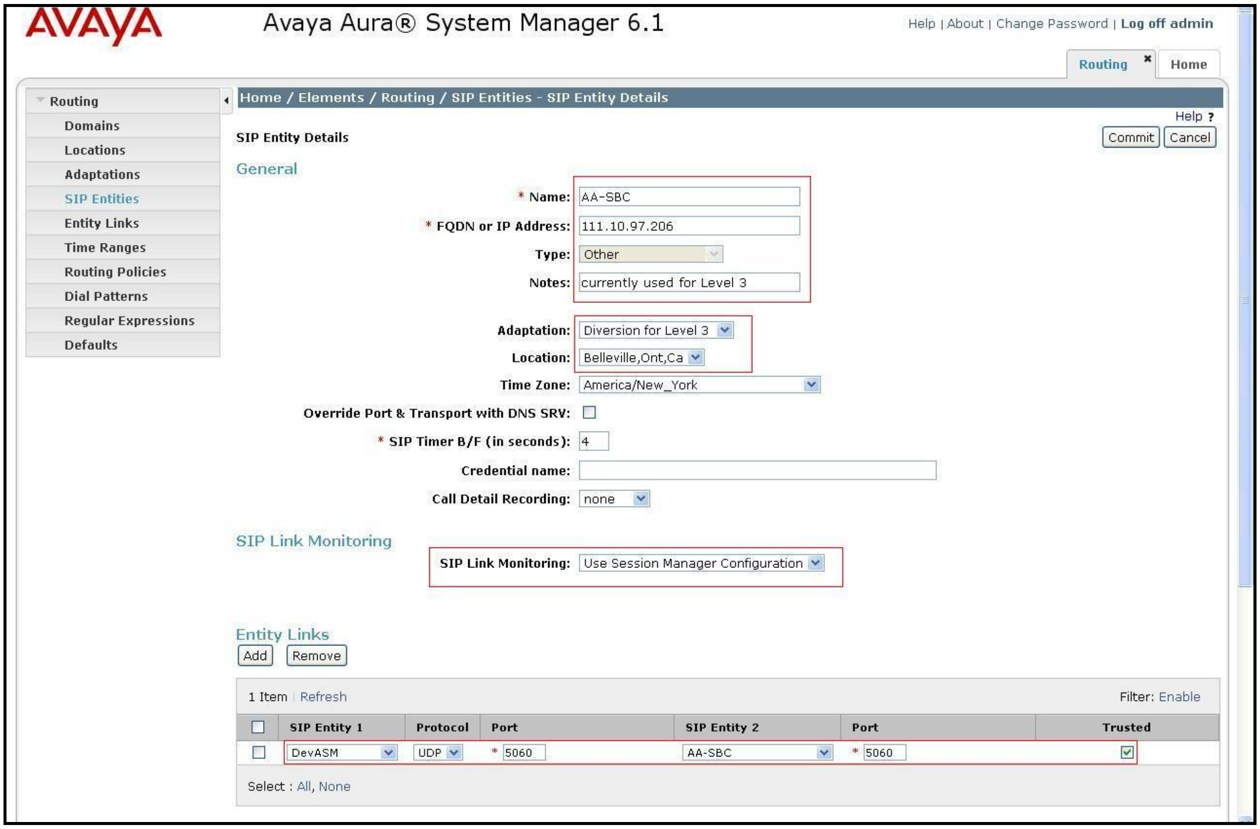


Figure 6.7 Adding SIP Entity for Avaya Aura® SBC

b) Click **Commit**.

Note: In the **Entity Links** configuration, the option "**Trusted**" is mandatory.

6.6. Create Routing Policy for inbound call

This section shows how to configure Session Manager to add a **Routing Policy** for inbound call from Level 3 to CS1000E. As part of the dialing plan configuration, the **Routing Policy** instructs the Session Manager to route the SIP call from PSTN to the CS1000E SIP Gateway to terminate.

The "**Time of Day**" setting defines the range to apply the **Routing Policy** during the day. In this testing, just simply select the default name **24/7**. It means the **Routing Policy** is always in effect.

Figure 6.7 shows policy **Level 3 – Cust 0** was created.

- **Name**: Level 3 – Cust 0
- **SIP Entity as Destination**: car1-cores1-Cust_0
- **Time of Day**: 24/7

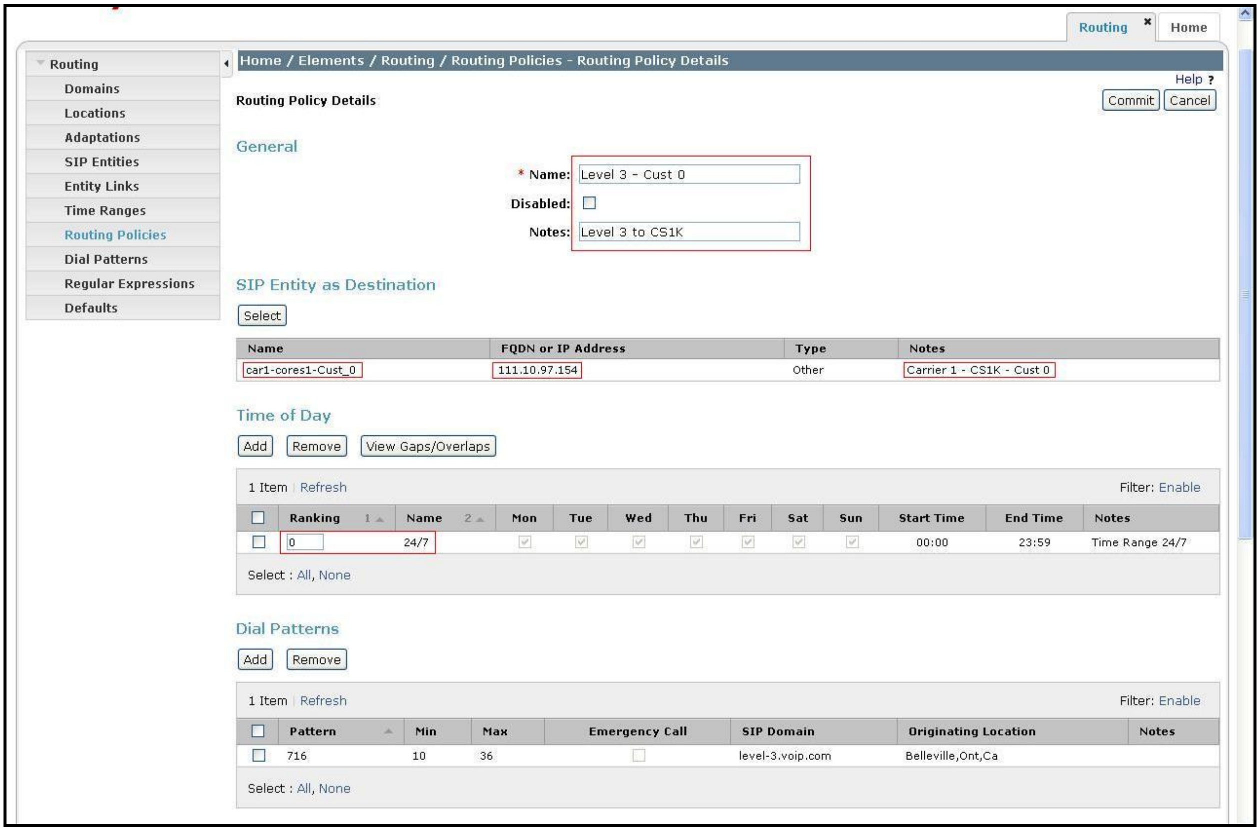


Figure 6.8 Adding Routing Policy for inbound call

6.7. Create Routing Policy for outbound call

Please refer to **Section 6.5** to create a **Routing Policy** for an outbound call. Based on the policy, the Session Manager routes the call from the CS1000E to the SIP Entity AA-SBC as the destination, then the Avaya Aura® SBC sends the request to Level 3.

Figure 6.9 shows policy **CS1000 – Cust 0 to Level 3** was created.

- **Name**: CS1000 – Cust 0 to Level 3
- **SIP Entity as Destination**: AA-SBC
- **Time of Day**: 24/7

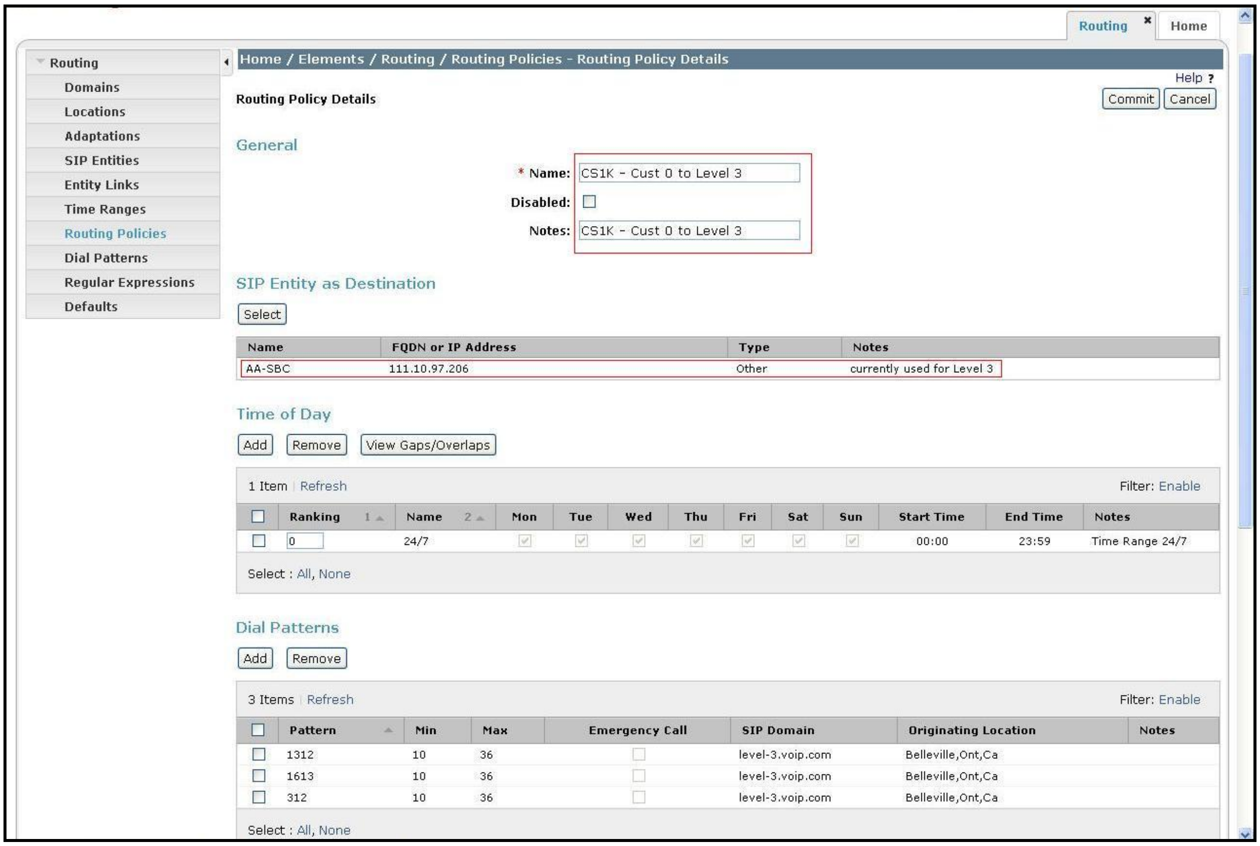


Figure 6.9 Adding Routing Policy for outbound call

6.8. Create Dial Pattern for inbound call

In this testing, Level 3 assigns DID numbers with prefix **716** to CS1000E. The DIDs are in 10 digits format. The Dial Pattern **716** on Session Manager is configured as an entry of Routing Policy **Level 3 – Cust 0**. It means when Session Manager receives inbound call with prefix **716**, it will routes the call to CS1000E SIP Gateway **car1-cores1-Cust_0** as the destination.
Figure 6.10 shows policy **Dial Pattern 716** was created.

a) In the **Introduction to Network Routing Policy** page (not shown), click **Dial Patterns** link on the left menu to open **Dial Patterns – Dial Pattern Details** page. Then click button **New** (not shown) to add a new Dial Pattern for inbound call with prefix **716**.

b) Under **Originating Locations and Routing Policy**, click **Add** (not shown). In the **Dial Patterns – Originating Locations and Routing Policy List** page (not shown), select **Originating Location** entry **Belleville,Ont,Ca** and **Routing Policies** entry **Level 3 – Cust 0**.

- **Pattern**: 716
- **Min**: 10 (digits)
- **Max**: 36 (default)
- **SIP Domain**: level-3.voip.com
- **Originating Location Name**: Belleville,Ont,Ca
- **Routing Policy Name**: Level 3 – Cust 0
- **Routing Policy Destination**: car1-cores1-Cust_0

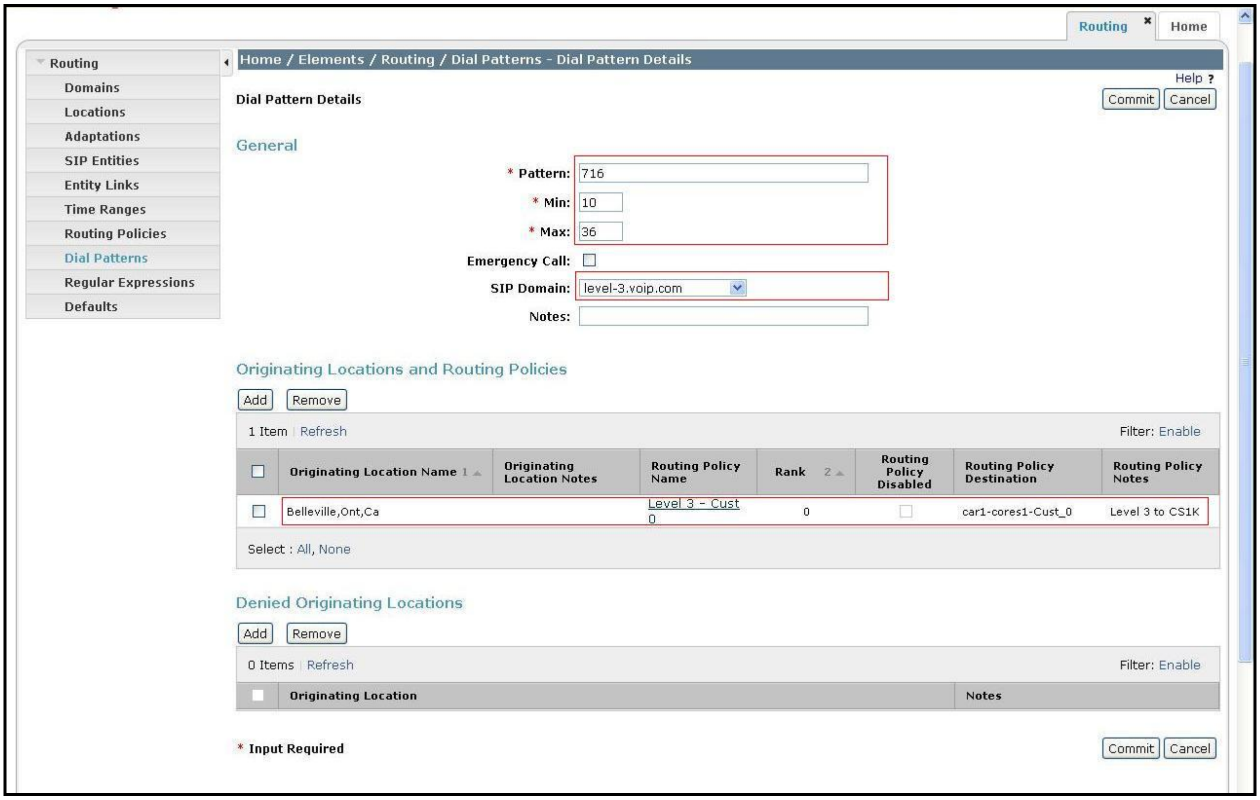


Figure 6.10 Adding Dial Pattern for inbound call

c) Click **Commit**.

6.9. Create Dial Pattern for outbound call

The **Dial Pattern** for outbound call is associated to the **Routing Policy CS1000 – Cust 0 to Level 3**. The **Dial Pattern** configuration on Session Manager has to match the dialing plan configure on CS1000E.

a) Dial Pattern with prefix **1**. For long distance calls, CS1000E sends 11 digits with prefix **1** to Level 3 system via Avaya Aura® SBC. To create the Dial Pattern **1**, the detail configuration is shown in
Figure 6.11.

- **Pattern**: 1
- **Min**: 11 (digits)
- **Max**: 36 (default)
- **SIP Domain**: level-3.voip.com
- **Originating Location Name**: Belleville,Ont,Ca
- **Routing Policy Name**: Level 3 – Cust 0
- **Routing Policy Destination**: AA-SBC

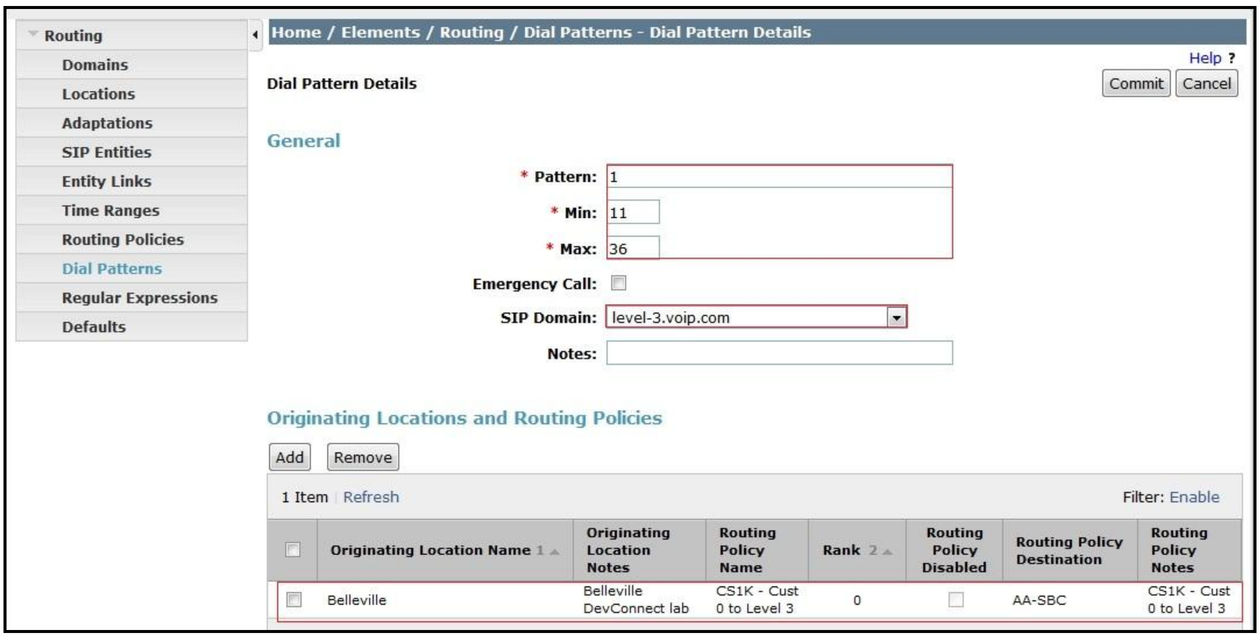


Figure 6.11 Adding Dial Pattern for outbound long distance call with prefix 1

b) Dial Pattern with prefix **613**. During testing, CS1000E sends 10 digits with prefix **613** defined by Level 3 to reach the PSTN phones in the Avaya lab. To create the **Dial Pattern 613**, the detail configuration is shown in
Figure 6.12.

- **Pattern**: 613

- **Min**: 10 (digits)
- **Max**: 36 (default)
- **SIP Domain**: level-3.voip.com
- **Originating Location Name**: Belleville,Ont,Ca
- **Routing Policy Name**: Level 3 – Cust 0
- **Routing Policy Destination**: AA-SBC

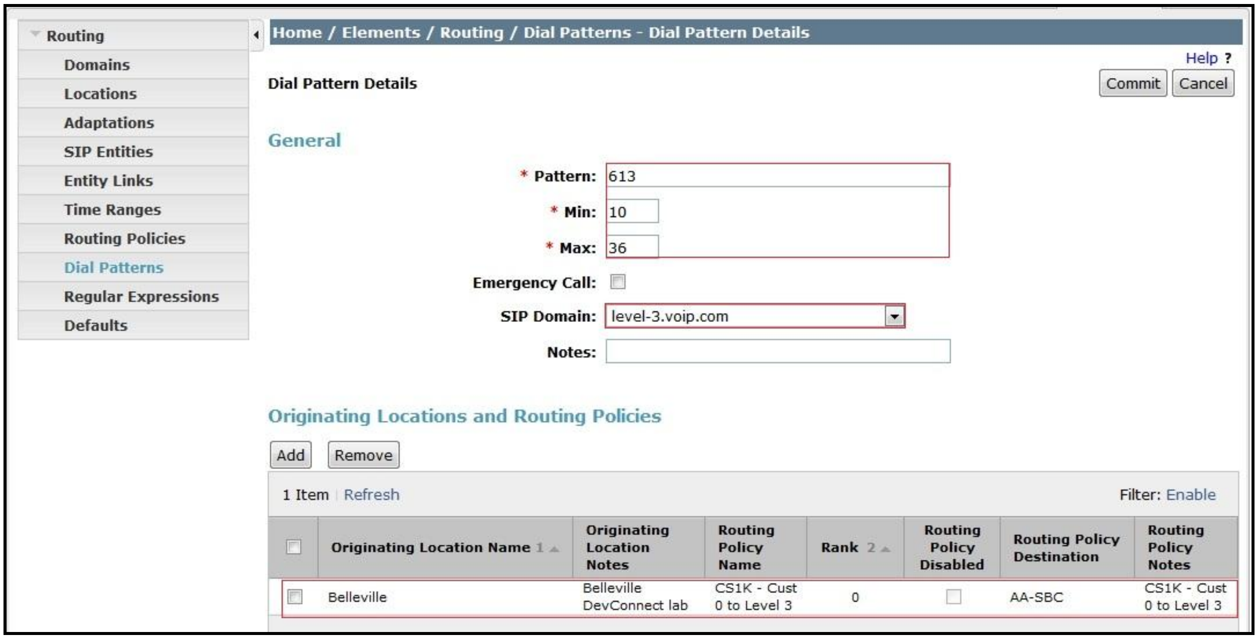


Figure 6.12 Adding Dial Pattern for outbound call with prefix 613

c) Dial Pattern with prefix **0**. CS1000E sends **0** or **0+10** digits to reach operator at Level 3. Level 3 also uses the same prefix **011** for the outbound international call. Thus, the Dial Pattern **0** should have flexible length. To create the Dial Pattern **0**, the detail configuration is shown in **Figure** 6.13.

- **Pattern**: 0
- **Min**: 1 (digits)
- **Max**: 36 (default)
- **SIP Domain**: level-3.voip.com
- **Originating Location Name**: Belleville,Ont,Ca
- **Routing Policy Name**: Level 3 – Cust 0
- **Routing Policy Destination**: AA-SBC

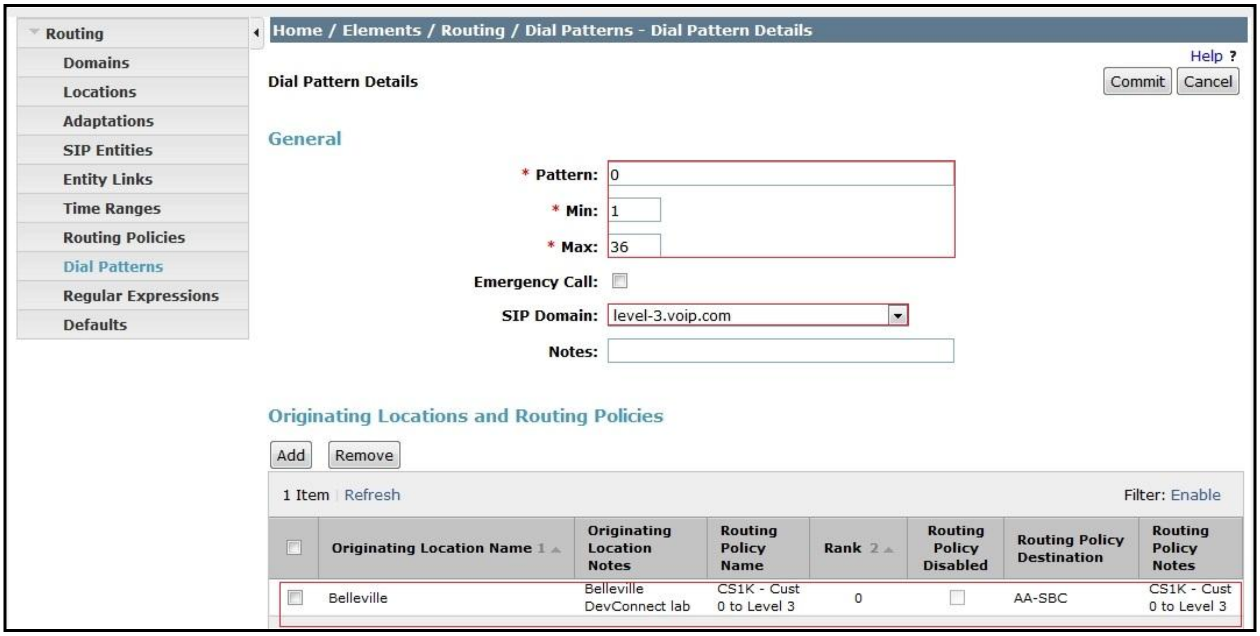


Figure 6.13 Adding Dial Pattern for outbound special call with prefix 0

d) Dial Pattern with prefix **411**. As part of the dialing plan, the **Dial Pattern 411** routes the call from CS1000E to 411 services hosted on Level 3. To create the Dial Pattern **411**, the detail configuration is shown in
Figure 6.14.

- **Pattern**: 411
- **Min**: 3 (digits)
- **Max**: 3 (default)
- **SIP Domain**: level-3.voip.com
- **Originating Location Name**: Belleville,Ont,Ca
- **Routing Policy Name**: Level 3 – Cust 0
- **Routing Policy Destination**: AA-SBC

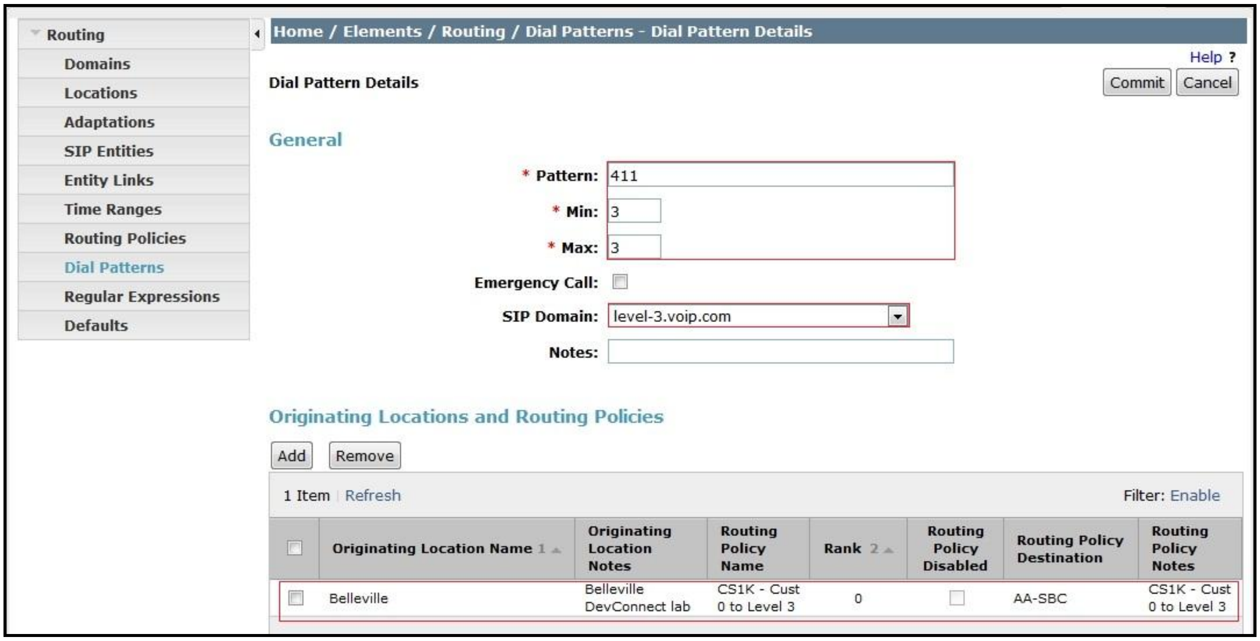


Figure 6.14 Adding Dial Pattern for outbound 411 calls

e) Dial Pattern with prefix **911**. As part of the dialing plan, the **Dial Pattern 911** routes the call from CS1000E to 911 emergency services hosted on Level 3. To create the Dial Pattern **911**, the detail configuration is shown in
Figure 6.15.

- **Pattern**: 911
- **Min**: 3 (digits)
- **Max**: 3 (default)
- **SIP Domain**: level-3.voip.com
- **Originating Location Name**: Belleville,Ont,Ca
- **Routing Policy Name**: CS1_To_Level3
- **Routing Policy Destination**: AA-SBC

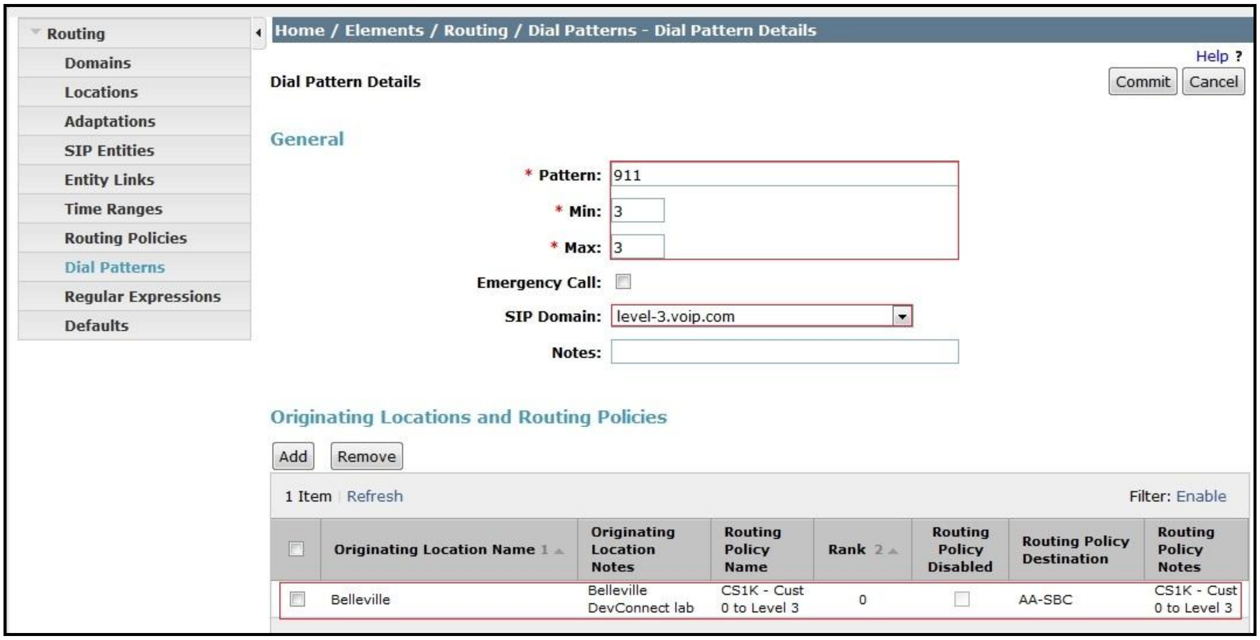


Figure 6.15 Adding Dial Pattern for outbound 911 calls

6.10.Add Adaptation Module

Session Manager can be configured with Adaptation modules that modify SIP messages before or after routing decisions have been made. There are 2 steps needed to convert the History-Info header sent by the CS1000E into the Diversion header as required by the Level 3 network for redirected calls. A generic Adaptation module **CS1000Adapter** is used to convert the History-Info header from the Nortel used format to the Avaya format of History-Info. The conversion changes the index and reasons to a Avaya standard format. A generic Adaptation module **DiversionTypeAdapter** is used to convert History-Info headers that are in Avaya format to Diversion Headers. The two adaptations take the CS1000E History-Info header and first converts it into an Avaya standard and then converts it into Diversion header. Other Adaptation modules are built on this generic module, and can modify other headers to permit interoperability with third party SIP products.

To view or change adaptations, select **Routing → Adaptations**. Click on the checkbox corresponding to the name of an adaptation and **Edit** to edit an existing adaptation, or the **New** button to add an adaptation. Click the **Commit** button after changes are completed.

The following screen shows a portion of the list of adaptations in the sample configuration.

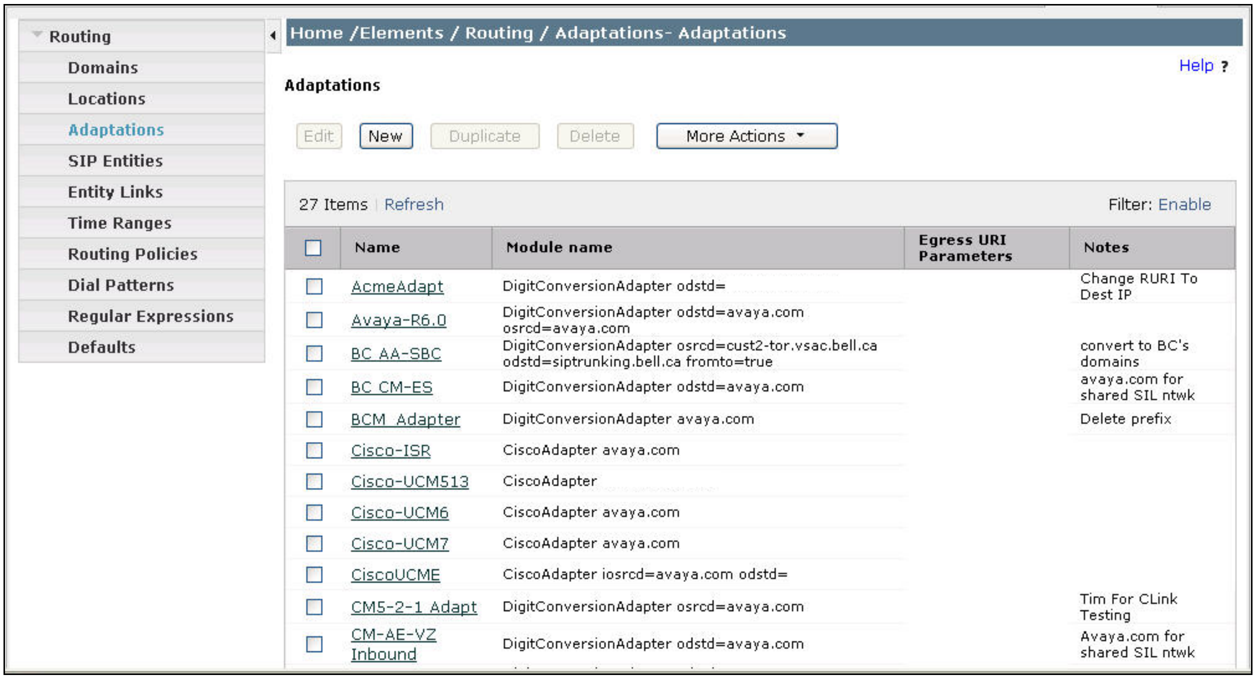


Figure 6.16 Adaptation list

CS1000Adapter – This adaptation is used to convert the History-Info header from the configuration used by Nortel into the configuration used by Avaya. The conversion involves changing the index format and the reason codes into the standard used by Avaya. To create an adaptation for the CS1000E History-Info header, click on the **New** button as shown in **Figure** 6.16. The configuration is detailed below and the final screen is shown in **Figure** 6.17.

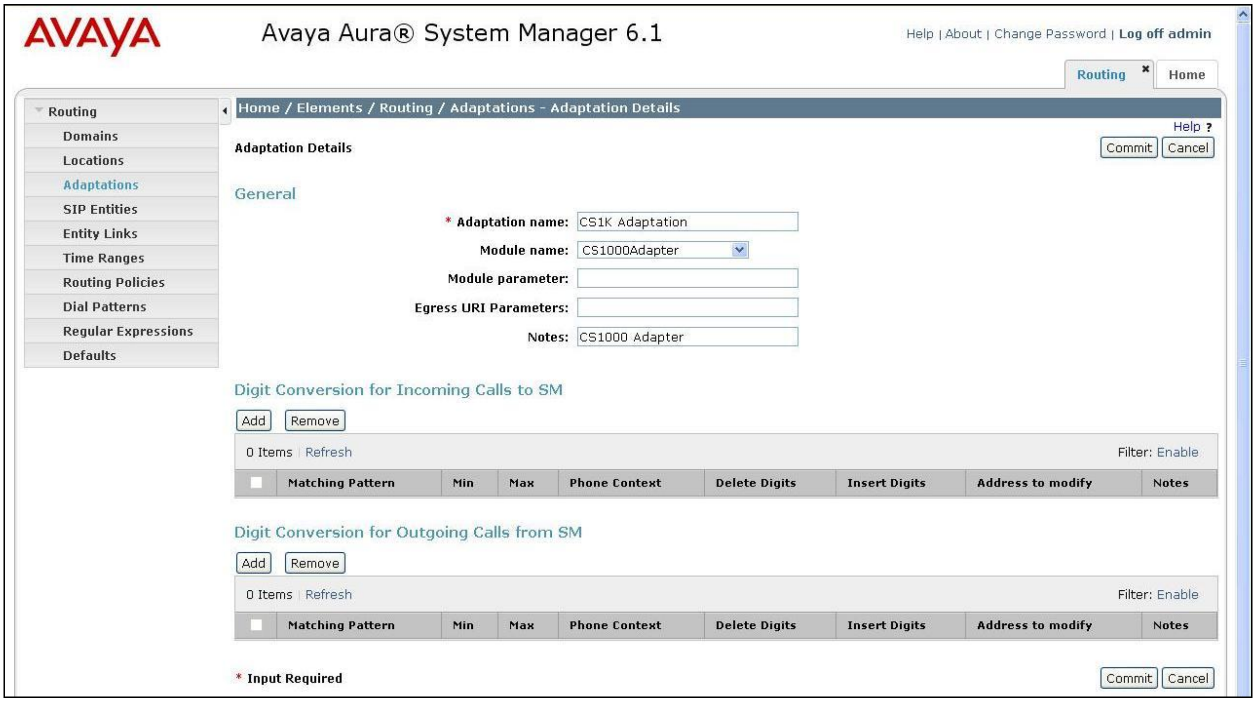


Figure 6.17 Adaptation for History-Info conversion

DiversionTypeAdapter – This adaptation is used to convert History-Info headers (which are not supported by the Level 3 Sip Trunk) sent by the CS1000E in certain outbound calls to Level 3, to Diversion headers. This is required for call scenarios such as Call Forwarding. To create an adaptation for the Diversion header, click on the **New** button as shown in **Figure** 6.16. The configuration is detailed below and the final screen is shown in **Figure** 6.18.

- **Adaptation name**: Diversion for Level 3 (or any meaningful name)
- **Module name**: DiversionTypeAdapter
- **Module parameter**: osrcd=333.55.35.85 odstd=111.10.98.104 MIME=no
- **Egress URI Parameters**: N/A
- **Notes**: Outbound diversion for Level3 (or any descriptive comment)

Module parameters can be added to the adaptation to perform further customizations

- **osrcd=333.55.35.85**. This configuration enables the outbound source domain to be overwritten with **333.55.35.85**. For example, for outbound PSTN calls from the Avaya CS1000E to Level 3, the PAI header will contain 333.55.35.85, the address of the border element, as expected by Level 3.
- **odstd=111.10.98.104**. This configuration enables the outbound destination domain to be overwritten with **111.10.98.104**. For example, for outbound PSTN calls from the Avaya CS1000E to Level 3, the Request-URI will contain **111.10.98.104** as expected by Level 3.
- **MIME=no**. This configuration is used to remove unnecessary CS1000E SIP headers and multipart SDP.

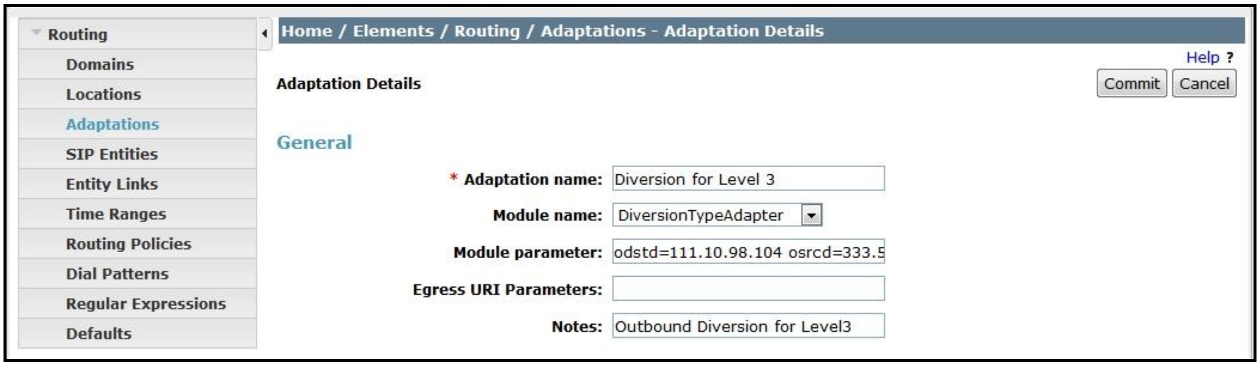


Figure 6.18 Adaptation for Diversion header

7. Configure Avaya Aura® Session Border Controller

The AA-SBC configuration is done in two parts. The first part is done during the AA-SBC installation via the installation wizard. These Application Notes will not cover the AA-SBC installation in its entirety. It is assumed that the first step on installing the System Platform along with the network settings has been completed.

The second part of the configuration is done after the installation is complete using the AA-SBC web interface. The resulting AA-SBC configuration file is shown in **Appendix A**.

7.1. Post Installation Configuration

The installation wizard configures the Session Border Controller for use with a generic service provider therefore additional manual changes need to be performed to customize the configuration for Level 3. These changes are performed by accessing the browser-based GUI of the AA-SBC, using the URL **https://<ip-address>**, where **<ip-address>** is the private IP address configured for the AA-SBC. Log in with proper credentials.

Figure 7.1 AA-SBC login

7.2. Options Frequency

To set the frequency of the OPTIONS messages sent from the Avaya Aura® SBC to the service provider, first navigate to **vsp → enterprise → servers → sig-gateway Telco**. Click **Show Advanced** in **Figure 7.2.**

Figure 7.2 Advance SIP-gateway to Telco

Scroll down to the **routing** section of the form. Enter the desired interval in the **ping-interval** field as shown in
Figure 7.3. Click **Set** at the top of the form as shown in **Figure 7.2**.

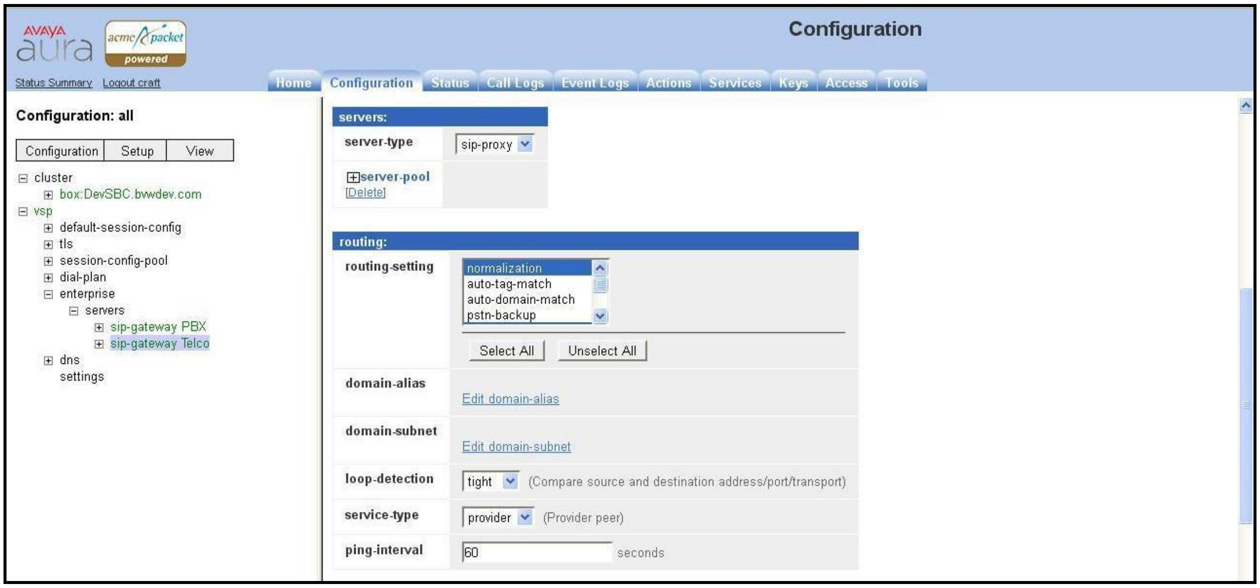


Figure 7.3 Options Frequency

Similar procedures can be used to set the Options Frequency from AA-SBC to Session Manager in **vsp → enterprise → servers → sig-gateway PBX**.

7.3. Blocked Headers

The P-Location and P-Site headers are sent in SIP messages from the Session Manager. The CS1000E sends P-Asserted-Identity, History-Info and x-nt-xxx. These headers should not be exposed external to the enterprise. For simplicity, these headers were simply removed (blocked) from both requests and responses for both inbound and outbound calls. To create a rule for blocking a header on an outbound call, first navigate to **vsp → session-config-pool → entry ToTelco → header-settings**. Click **Edit blocked-header** link on the right panel shown in **Figure** 7.4.

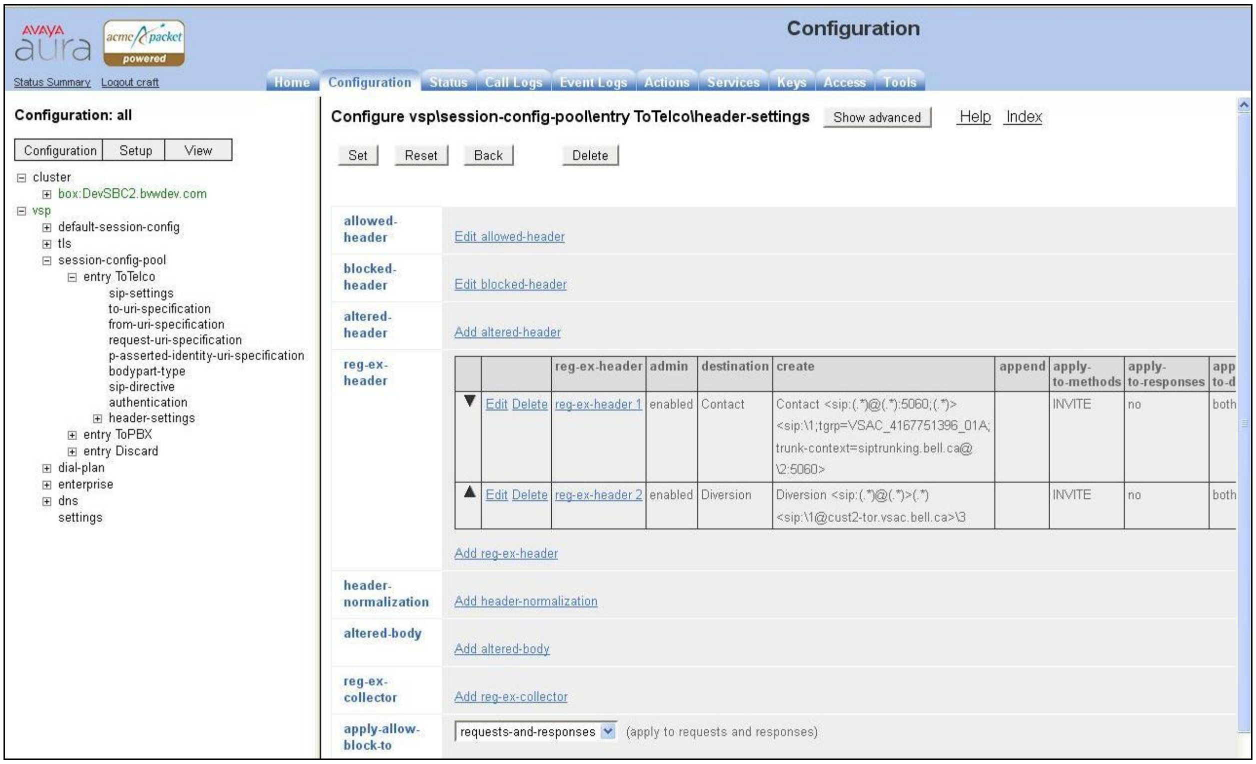


Figure 7.4 Header Settings

In the right pane that appears, click **Add.** In the blank field that appears, enter the name of the header to be blocked. After all the blocked headers are added, click **OK** to continue. **Figure** 7.5 shows the **P-Location** and the **P-Site** headers were configured to be blocked.

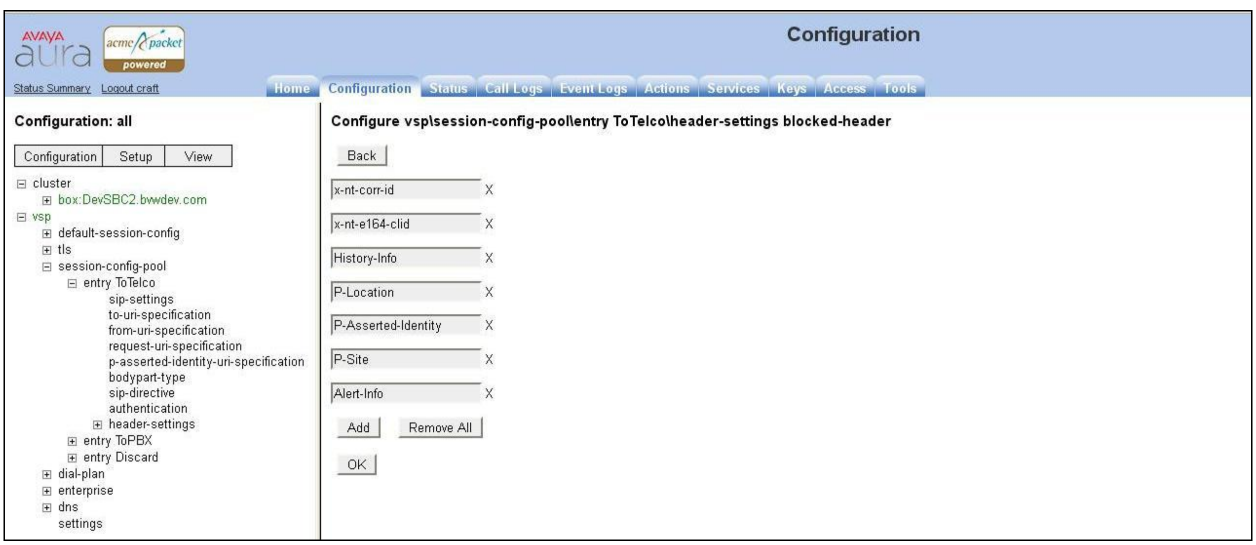


Figure 7.5 Blocked Header Configuration

The list of blocked-headers will appear in the right pane as shown in

Figure 7.6. Click **Set** to complete this configuration.

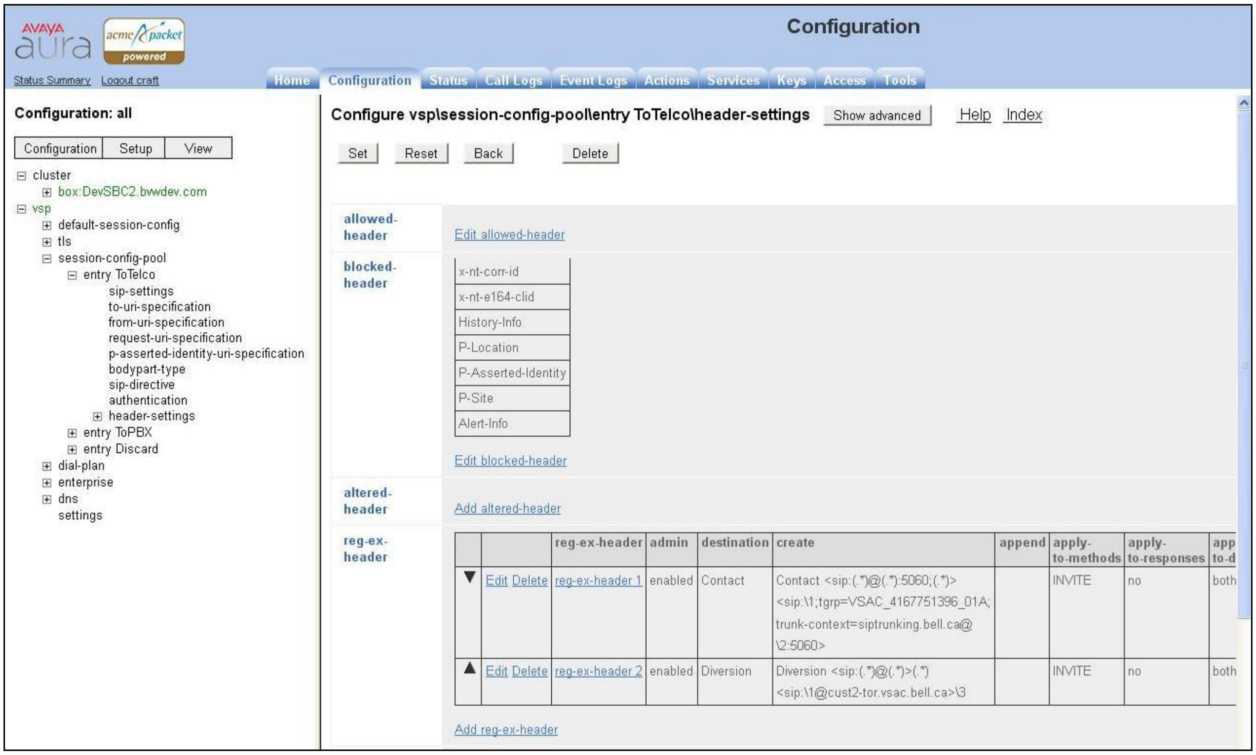


Figure 7.6 Blocked Headers

7.4. Allow Header Modification

Level 3 does not support the UPDATE method and as such it needs to be removed in both the outbound and inbound directions from the Allow header. The following steps outline the procedure to create a header manipulation rule to make the necessary changes.

Navigate to **vsp → session-config-pool → entry ToTelco → header-settings**. On the screen displayed in
Figure 7.6 click the **Add altered-header** link in the right panel.

Figure 7.7 shows the edit screen for a previously added altered-header.

In the **number** field, enter an appropriate unused number. Since this is the first altered-header rule, number 0 was used. In the **source-header** field, enter **Allow**. In the source-field area:

- in the **expression** field, either enter a value to match directly, or click the **regular expression** link for assistance in creating the proper value. In the sample configuration, the rule will match on, UPDATE appearing in the Allow header.
- in the **replacement** field, enter simply **\1**, which will be the entire Allow header extracted with the (.*) in the expression field earlier. In effect this will remove UPDATE from the Allow header.

Retain the default settings for other configuration fields. Click the **Create** button (not shown) if adding a new altered-header; click the **Set** button if editing an existing altered-header.

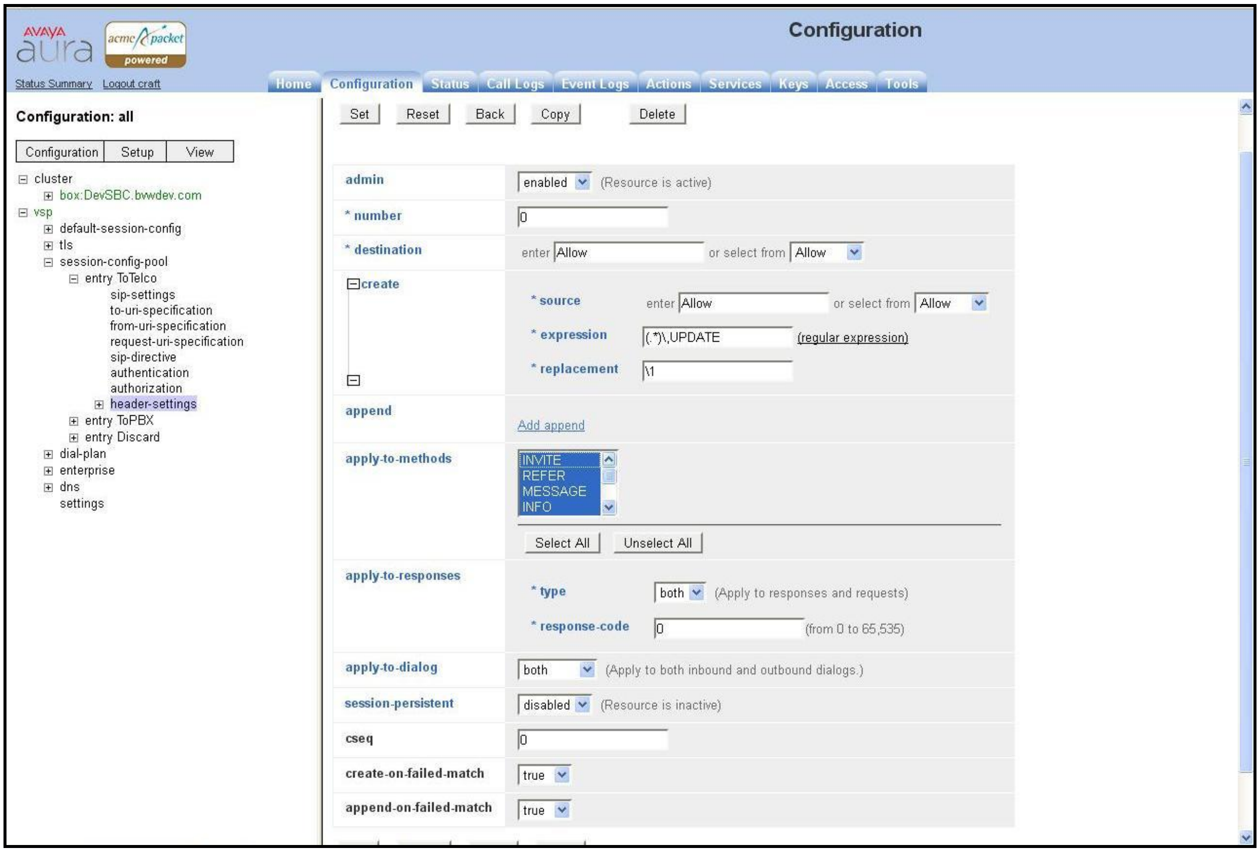


Figure 7.7 UPDATE reg-ex-header configuration

Figure 7.8 shows a summary of the header manipulations for SIP messages sent to the Level 3 network, including the blocked headers and any modified headers.

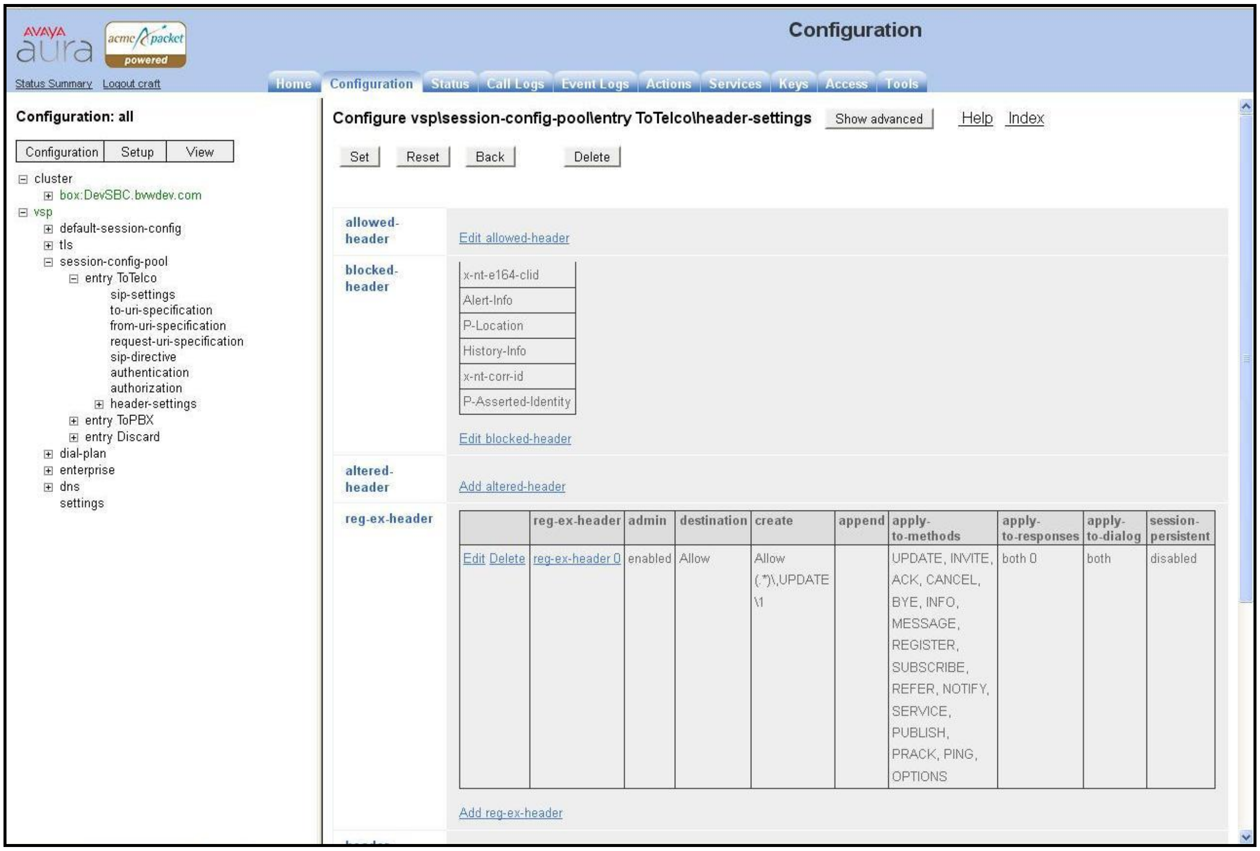


Figure 7.8 Header Manipulations

The same configuration is duplicated for the ToPBX direction. In this way the UPDATE method is also stripped off the incoming Allow headers sent by Level 3. The blocked headers are technically not needed but are added for completeness. The same steps are followed as detailed above but for the ToPBX direction.

Figure 7.9 shows a summary of the header manipulations for SIP messages sent to the Avaya network, including the blocked-headers and reg-ex-header.

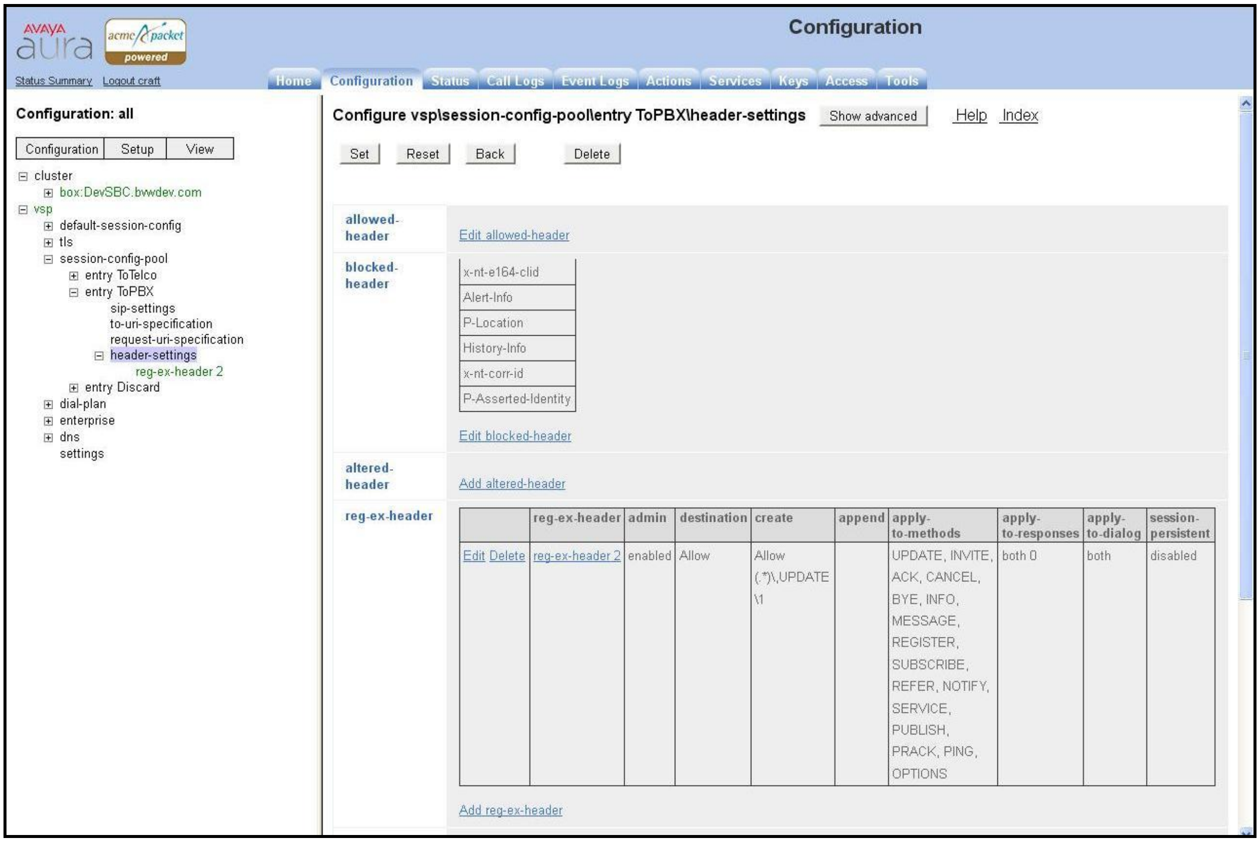


Figure 7.9 toPBX Header settings

7.5. Third Party Call Control

Disable third party call control. Navigate to **vsp → default-session-config → third-party-call-control**. Set the **admin** field to **disabled**.

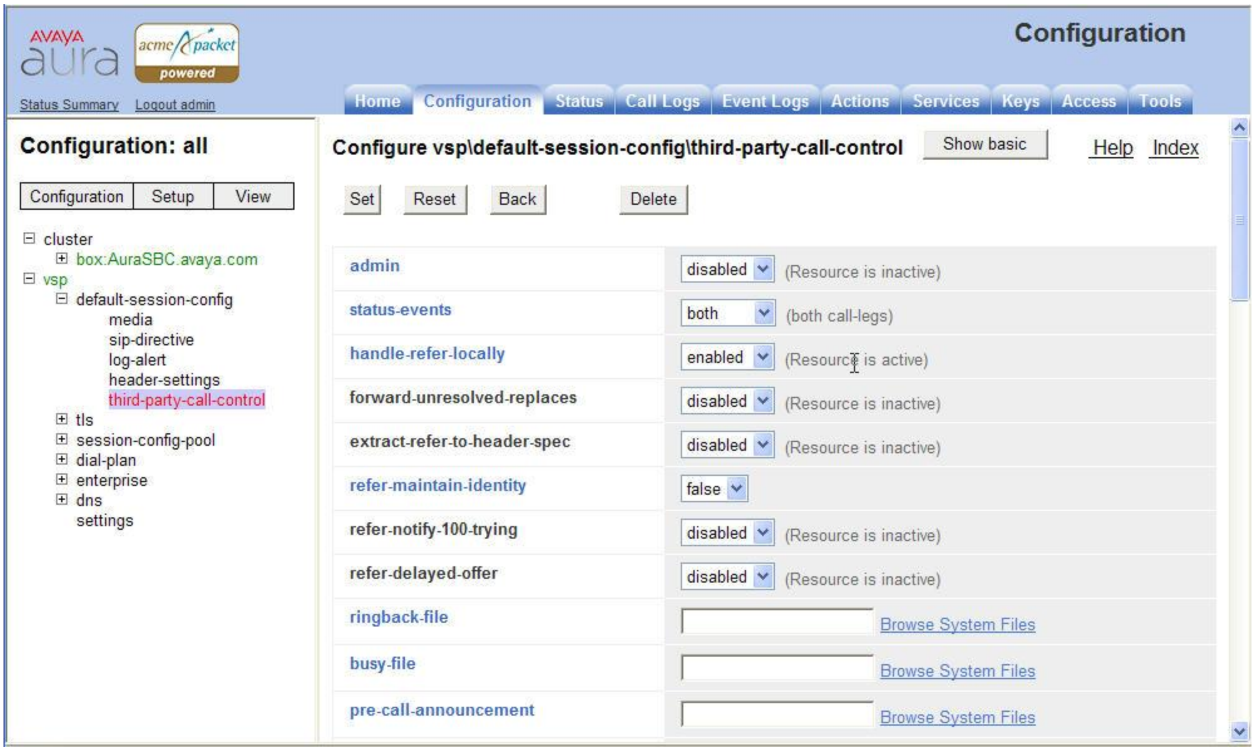
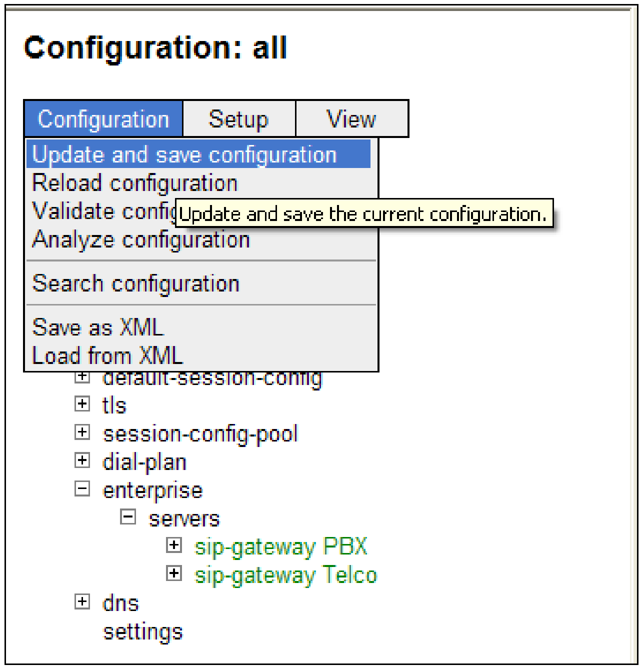


Figure 7.10 third party call control

7.6. Save the Configuration

To save the configuration, begin by clicking on **Configuration** in the left pane to display the configuration menu. Next, select **Update and save configuration**.



7.11 Save Configuration

8. Verification Steps

The following steps may be used to verify the configuration.

8.1. General

Place an **inbound/ outbound** call from/ to a PSTN phone to/ from an internal CS1000E phone, answer the call, and verify that two-way speech path exists. Check call display name and number to ensure the correct info was sent/ received. Perform hold/ retrieve. Verify the call remains stable for several minutes and disconnect properly.

8.2. Verify Call Establishment on CS1000E Call Server

a) Active Call Trace (LD 80)
The following is an example of one of the commands available on the CS1000E to trace the DN when the call is in progress and or idle. The call scenario involved the CS1000E extension 1215 calling PSTN phone number 6139675204.

- Login 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 0 1215**.
- After call is released, issue command **trac 0 1215** 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 1215 is in an active call:

```
>ld 80

.trac 0 1215

ACTIVE  VTN 096 0 00 01

ORIG   VTN 096 0 00 01  KEY 0  SCR MARP  CUST 0  DN 1215  TYPE 2004P1
  SIGNALLING ENCRYPTION: INSEC
  MEDIA ENDPOINT IP: 111.10.98.40  PORT: 5200
TERM   VTN 100 0 00 31   VTRK IPTI  RMBR  100 32 OUTGOING VOIP GW CALL
  FAR-END SIP SIGNALLING IP: 111.10.97.184
  FAR-END MEDIA ENDPOINT IP: 111.10.97.184  PORT: 20004
  FAR-END VendorID: AVAYA-SM-6.1.1.0.611023
MEDIA PROFILE: CODEC G.711 MU-LAW  PAYLOAD 20 ms  VAD OFF
RFC2833:  RXPT  101   TXPT  101   DIAL DN 616139675204
MAIN_PM  ESTD
TALKSLOT  ORIG  2   TERM  39
EES_DATA:
NONE
QUEU  NONE
CALL ID 0 34360


----  ISDN ISL CALL (TERM) ----
```

```
CALL REF # =  416
BEARER CAP =  VOICE
HLC =
CALL STATE =  10     ACTIVE
CALLING NO =  7162611215  NUM_PLAN:E164    TON:NATIONAL  ESN:NPA
CALLED NO  =  16139675204  NUM_PLAN:PRIVATE     TON:NETWORK SPECIFIC  ESN:SPN
```

This is the example after the call on 1215 is completed.

```
.trac 0 1215

IDLE VTN 096 0 00 01   MARP
```

b) SIP Trunk monitoring (LD 32)
Place a call inbound from PSTN (6139675204) to CS1000E (7162611215). Then check the SIP Trunk status by using LD 32, the output below shows that one trunk is busy.

```
>ld 32
NPR003
.stat 100 0
031 UNIT(S) IDLE
001 UNIT(S) BUSY
000 UNIT(S) DSBL
000 UNIT(S) MBSY
```

And this is the example after the call is completed, shows that there are no trunks busy.

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

8.3. Protocol Traces

Below are Wireshark traces of the same call scenario that has been made in **Section 8.2**.

The following SIP headers are inspected:

- RequestURI: verify the request number and either SIP domain
- From: verify the display name and display number.
- To: verify the display name and display number.
- Diversion: verify the call forward information and reason code.
- Remote-Party-ID: verify the display name; display number and privacy settings.

The following attributes in SIP message body are inspected:

- Connection Information (c): verify IP address of far end endpoint
- Time Description (t): verify session timeout of far end endpoint
- Media Description (m): verify audio port, codec, DTMF event description
- Media Attribute (a): verify specific audio port, codec, ptime, send/ receive ability, DTMF event and fax attributes.

a) SIP/INVITE from CS1000E captured at AA-SBC OUTSIDE interface.

INVITE sip:13129574679@333.55.35.85 SIP/2.0
From: "Level3 2001" <sip:7162611215@111.10.98.104>;tag=68620a87-13c4-4ef0e466-f350adc7-3efe95d0
To: <sip:13129574679@333.55.35.85>
Call-ID: CXC-55-59a643d0-68620a87-13c4-4ef0e466-f350adc7-12f0a682@111.10.98.104
CSeq: 1 INVITE
Via: SIP/2.0/UDP 111.10.98.104:5060;branch=z9hG4bK-214cb-4ef0e466-f350adc8-12eb59c9
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
Remote-Party-ID: "Level3 2001" <sip:7162611215@333.55.35.85;user=phone>;party=calling;screen=no;privacy=off
Max-Forwards: 65
Contact: <sip:7162611215@111.10.98.104:5060;maddr=111.10.98.104;transport=udp>
Allow: INVITE,ACK,BYE,REGISTER,REFER,NOTIFY,CANCEL,PRACK,OPTIONS,INFO,SUBSCRIBE
Content-Type: application/sdp
Content-Length: 263

v=0
o=- 99 1 IN IP4 111.10.98.104
s=-
c=IN IP4 111.10.98.104
t=0 0
m=audio 22322 RTP/AVP 0 8 18 101 111
c=IN IP4 111.10.98.104
a=rtpmap:101 telephone-event/8000
a=rtpmap:111 X-nt-inforeq/8000
a=fmtp:18 annexb=no
a=fmtp:101 0-15
a=ptime:20
a=sendrecv

b) SIP/401 from Level 3 for Authentication challenge.

SIP/2.0 401 Unauthorized

From: "Level3 2001" <sip:7162611215@111.10.98.104>;tag=68620a87-13c4-4ef0e466-f350adc7-3efe95d0
To: <sip:13129574679@333.55.35.85>;tag=SDf2h1a99-482934549-1324410068328
Call-ID: CXC-55-59a643d0-68620a87-13c4-4ef0e466-f350adc7-12f0a682@111.10.98.104
CSeq: 1 INVITE
Via: SIP/2.0/UDP 111.10.98.104:5060;branch=z9hG4bK-214cb-4ef0e466-f350adc8-12eb59c9
WWW-Authenticate: DIGEST qop="auth",nonce="BroadWorksXgwfbk50oT9fb1f3BW",algorithm=MD5,realm="BroadWorks"
Content-Length: 0

c) SIP/INVITE from CS1000E with Digest Authentication.

INVITE sip:13129574679@333.55.35.85 SIP/2.0
From: "Level3 2001" <sip:7162611215@111.10.98.104>;tag=68620a87-13c4-4ef0e466-f350adc7-3efe95d0
To: <sip:13129574679@333.55.35.85>
Call-ID: CXC-55-59a643d0-68620a87-13c4-4ef0e466-f350adc7-12f0a682@111.10.98.104
CSeq: 2 INVITE
Via: SIP/2.0/UDP 111.10.98.104:5060;branch=z9hG4bK-214cc-4ef0e467-f350ae17-5246c985
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
Remote-Party-ID: "Level3 2001" <sip:7162611215@333.55.35.85;user=phone>;party=calling;screen=no;privacy=off
Max-Forwards: 65
Authorization: Digest username="1-23Q-3413", realm="BroadWorks", nonce="BroadWorksXgwfbk50oT9fb1f3BW", uri="sip:13129574679@level-3.voip.com;user=phone", response="f67fcb0a4cb805869e623ba9af1064fe", algorithm=MD5, cnonce="1ca18421", qop=auth, nc=00000001
Contact: <sip:7162611215@111.10.98.104:5060;maddr=111.10.98.104;transport=udp>
Allow: INVITE,ACK,BYE,REGISTER,REFER,NOTIFY,CANCEL,PRACK,OPTIONS,INFO,SUBSCRIBE
Content-Type: application/sdp
Content-Length: 263

v=0
o=- 99 1 IN IP4 111.10.98.104
s=-
c=IN IP4 111.10.98.104
t=0 0
m=audio 22322 RTP/AVP 0 8 18 101 111
c=IN IP4 111.10.98.104
a=rtpmap:101 telephone-event/8000

a=rtpmap:111 X-nt-inforeq/8000
a=fmtp:18 annexb=no
a=fmtp:101 0-15
a=ptime:20
a=sendrecv

d) SIP/200OK from Level 3.

SIP/2.0 200 OK
From: "Level3 2001" <sip:7162611215@111.10.98.104>;tag=68620a87-13c4-4ef0e466-f350adc7-3efe95d0
To: <sip:13129574679@333.55.35.85>;tag=SDf2h1a99-388586082-1324410068668
Call-ID: CXC-55-59a643d0-68620a87-13c4-4ef0e466-f350adc7-12f0a682@111.10.98.104
CSeq: 3 OPTIONS
Via: SIP/2.0/UDP 111.10.98.104:5060;branch=z9hG4bK-214cd-4ef0e467-f350af6c-34ff4ee8
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Accept: multipart/mixed,application/media_control+xml,application/sdp
Supported:
Allow-Events: call-info,line-seize,dialog,message-summary,as-feature-event
Content-Length: 0

9. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Communication Server 1000 7.5, Avaya Aura® Session Manager 6.1 and Avaya Aura® Session Border Controller 6.0.2 to the Level 3 SIP Trunking service.

All of the test cases have been executed. Noting the observations seen during testing as described 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 Avaya Aura® Session Border Controller 6.0.2.

10. Additional 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] *Avaya Aura® Session Border Controller System Administration Guide, V.6.0, September 2010*

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.