Avaya Solution & Interoperability Test Lab

Applications Notes for Avaya Communication Server 1000E Release 7.6 with Avaya Aura® Session Manager 6.3 and Avaya Session Border Controller for Enterprise 6.2 with AT&T IP Toll Free SIP Trunk Service – Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Aura® Session Manager 6.3, Avaya Communication Server 1000E 7.6 and Avaya Session Border Controller for Enterprise 6.2 with the AT&T IP Toll Free service using **AVPN** or **MIS/PNT** transport connections.

Avaya Aura® Session Manager 6.3 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Communication Server 1000E 7.6 is a telephony server, and is the point of connection between the enterprise endpoints and Avaya Aura® Session Manager. Avaya Session Border Controller for Enterprise 6.2 is the point of connection between Avaya Aura® Session Manager and the AT&T IP Toll Free service and is used to not only secure the SIP trunk, but also to make adjustments to the SIP signaling for interoperability. In addition, Avaya Aura® Contact Center is used to provide Agent access for Avaya Communication Server 1000E

The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks. Note that these Application Notes do NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service.

AT&T is a member of Avaya DevConnect Service Provider program. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program.

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1. Introduction

These Application Notes describe the steps for configuring Avaya Aura® Session Manager 6.3 (referred to in subsequent sections of this document as Session Manager), Avaya Communication Server 1000E 7.6 (referred to in subsequent sections of this document as CS1000E), and Avaya Session Border Controller for Enterprise 6.2 (referred to in subsequent sections of this document as Avaya SBCE) with the AT&T IP Toll Free service using **AVPN** or **MIS/PNT** transport connections.

Avaya Aura® Session Manager 6.3 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura® Session Manager is provisioned using the Avaya Aura® System Manager 6.3.5 platform (referred to in subsequent sections of this document as System Manager). Avaya Communication Server 1000E 7.6 is a telephony application server and is the point of connection between the enterprise endpoints and Avaya Aura® Session Manager.

In addition, Avaya Call Pilot® is used in conjunction with the Avaya Communication Server 1000E to provide voice mail access, as well as Avaya Aura® Contact Center which provide Agents access functionality. While both of these platforms are discussed in the following sections, their provisioning is beyond the scope of this document.

An Avaya Session Border Controller for Enterprise 6.2 is the point of connection between Avaya Aura® Session Manager and the AT&T IP Toll Free service and is used to not only secure the SIP trunk, but also to make adjustments to the SIP signaling for interoperability. In addition, Avaya Aura® Contact Center is used to provide Agent access for Avaya Communication Server 1000E

The AT&T IP Toll Free service is a managed Voice over IP (VoIP) communications solution that provides toll-free services over SIP trunks utilizing AVPN or MIS/PNT[1] transport.

Note - These Application Notes do NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service. **That solution is *not* supported by the CS1000E.**

2. General Test Approach and Test Results

The test environment consisted of:

- A simulated enterprise with System Manager, Session Manager, CS1000E, Avaya 11xx phones (UniStim and SIP), fax machines (Ventafax application), Avaya SBCE, and Avaya Call Pilot® voice messaging.
- A laboratory version of the AT&T IP Toll Free service, to which the simulated enterprise was connected via AVPN transport.

[1] MIS/PNT transport does not support compressed RTP (cRTP), however AVPN transport does support cRTP..

2.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying inbound call flows (see **Section 3.2** for examples) between Session Manager, CS1000E, Avaya SBCE, and the AT&T IP Toll Free service.

The compliance testing was based on a test plan provided by AT&T, for the functionality required for certification as a solution supported on the AT&T network. Calls were made from the PSTN across the AT&T IP Toll Free service network. The following features were tested as part of this effort:

- SIP trunking.
- T.38 Fax.
- Passing of DTMF events and their recognition by navigating automated menus.
- PBX and AT&T IP Toll Free service features such as hold, resume, conference and transfer.
- AT&T IP Toll Free features such as Legacy Transfer Connect and Alternate Destination Routing were also tested.

2.2. Test Results

The test objectives stated in below, with limitations as noted in **Section 2.2.1,** were verified.

1. Inbound AT&T IP Toll Free service calls to CS1000E telephones and Agents.
2. Call and two-way talk path establishment between PSTN and CS1000E telephones/Agents via the AT&T Toll Free service.
3. Basic supplementary telephony features such as hold, resume, transfer, and conference.
4. G.729 and G.711 codecs.
5. T.38 and G.711 fax calls from the AT&T IP Toll Free service/PSTN to CS1000E G3 and SG3 fax endpoints.
6. DTMF tone transmission using RFC 2833 between CS1000E and the AT&T IP Toll Free service/PSTN automated access systems.
7. Inbound AT&T IP Toll Free service calls to CS1000E that is directly routed to stations, and if unanswered, can be covered to Avaya Call Pilot®.
8. Requests for privacy (i.e., caller anonymity) for inbound calls to CS1000E from the PSTN, were verified.
9. SIP OPTIONS monitoring of the health of the SIP trunk was verified.
10. Long duration calls.
11. AT&T IP Toll Free features such as Legacy Transfer Connect and Alternate Destination Routing

2.2.1. Known Limitations

1. **CS1000E responds with ptime:10 in response to maxptime:30** – AT&T sends INVITEs with the SIP parameter *maxptime:30*. In response, CS1000E will send *ptime:10* for any UNIStim or Digital stations. This is a known issue. The AT&T AVPN transport service recommends the use of *ptime:30* for best bandwidth utilization. An Avaya SBCE script is

used to change the *maxptime:30* parameter to *ptime:30*, thereby making CS1000E respond with *ptime:30* as recommended (see **Sections 5.3.1, 5.3.2, and 8.3.9**).

2. **Removal of SIP Headers** – Depending on the call flow and the endpoints involved, the CS1000E and/or Session Manager may send multiple SIP headers that are not used by AT&T. In addition the AT&T IP Toll Free network does not support the History-Info header. Therefore in the interest of reducing packet overhead, the following headers are removed:
    - MIME type headers are removed by Session Manager Adaptations (see **Section 6.3.2**).
    - The Avaya SBCE is configured to remove the following SIP headers (see **Section 8.4.3**):
        - Alert-Info, x-nt-e164-clid, History-Info, Remote-Party-ID, Resource-Priority, AV-Global-Session-ID, P-AV-Message-ID, and P-Location.
3. **Telephone Events 101 and 111** - The CS1000E uses Telephone Event type 101 by default. This value is changed to the AT&T recommended value of 100 in the CS1000E (see **Section 5.9**). In addition, Telephone event type 111 is also sent by the CS1000E. This value is removed by the Avaya SBCE (see **Section 8.3.9**).
    - Note that the 1140E SIP telephones use a value of 101 for their RFC2833 Telephone Event Type, however no issues were found when 101 was used.
4. **G.726 codec is not supported by CS1000E**.
5. **The CS1000E may not populate the PAI header correctly in response to inbound calls.** In the reference configuration, the AT&T IP Toll Free service sends specific DNIS numbers in the R-URI sent to the Customer Premises Equipment (CPE). The AT&T IP Toll Free service prefers that the PAI and Contact headers sent in any subsequent CPE responses or requests contain these DNIS numbers as well. In addition, while the AT&T IP Toll Free service sends unique DNIS numbers in the INVITE Request URI (R-URI), it will send the same customer billing number in all TO headers. The CS100E Incoming Digit Translation table (IDT) is used to convert these DNIS digits to their associated local extensions. During testing it was found that the CS1000E may populate subsequent PAI headers with the associated destination extension, instead of the desired DNIS digits.
    - The workaround is to have the Avaya SBCE copy the DNIS number from the R-URI and insert it into the TO header prior to sending the INVITE to Session Manager/CS1000E, (the Avaya SBCE reinserts the billing number into the TO header prior to sending the messages back to AT&T). As a result, the CS1000E will populate the PAI with the associated AT&T IP Toll Free DNIS number.
    - It was also found that for inbound calls placed directly to the Call Pilot® main extension, the associated DNIS number in the R-URI/TO headers must also be defined in Call Pilot® as a Service Directory Number (see **Section 5.10**).
        - A CS1000E MR has been opened.
6. **Avaya SBCE support of T.38 –** During testing it was found that the current Avaya SBCE GA load (6.2 Q58) had an issue with T.38 fax. As a result, the use of Avaya SBCE load 6.2 Q48 is recommended at this time, if T.38 fax is required (see **Section 4**).
    - An Avaya SBCE MR was opened.

7. **Fax support** - G.711 and T.38 fax is supported (see **Item 6** above), and the sender and receiver of a fax call may use either Group 3 or Super Group 3 fax machines. However the T.38 fax protocol carries all fax transmissions as Group 3. Fax speeds of 14400, with Error Correction Mode, were observed in the reference configuration.

2.3. Support

2.3.1. AT&T

AT&T customers may obtain support for the AT&T IP Toll Free service by calling (800) 325-5555.

2.3.2. Avaya

Avaya customers may obtain documentation and support for Avaya products by visiting http://support.avaya.com. In the United States, (866) GO-Avaya (866-462-8292) provides access to overall sales and service support menus.

3. Reference Configuration

The reference configuration used in these Application Notes is shown in **Figure 1** and consists of the following:

- The CS1000E system provides the voice communications services for the enterprise site. The system is comprised of:
  - The MG1000E Gateway containing:
    - Call Server (CPPM).
    - Media Gateway Controller (MGC), which provides Digital Signaling Processor (DSP) resources.
    - Meridian Integration Recorded Announcement (MIRAN) card used for Music on Hold.
    - Avaya Call Pilot® messaging application.
  - IBM 306M **C**onsumer **O**ff the **S**helf (COTS) servers, COTS1 and COTS2.
    - Signaling Server and SIP Gateway (COTS1).
    - SIPLINE and UCM (COTS2).

Note – Only CS1000E system provisioning providing SIP trunk functionality is described in these application notes. For additional CS1000E system provisioning documentation, see **Section 11.**

- Avaya "desk" phones are represented with Avaya 1140E UNIStim IP, 1140E SIP, and Digital M3904 telephones.
- Session Manager provides core SIP routing and integration services that enables communication between disparate SIP-enabled entities, e.g., PBXs, SIP proxies, gateways, adjuncts, trunks, applications, etc. across the enterprise. Session Manager may use SIP over UDP, TCP, or TLS to communicate with SIP network elements. In the

reference configuration, Session Manager uses SIP over TCP to communicate with the Avaya SBCE, and SIP over TCP to communicate with the CS1000E.

- System Manager provides a common administration interface for centralized management of all Session Manager instances in an enterprise.
- Avaya SBCE provides address translation and SIP header manipulation between the AT&T IP Toll Free service and the enterprise internal network. TCP transport protocol is used between Avaya SBCE and Session Manager. UDP transport protocol is used between Avaya SBCE and the AT&T IP Toll Free service.
- An existing Avaya Call Pilot® system provides the corporate voice messaging capabilities in the reference configuration. **Note -** The provisioning of Avaya Call Pilot® is beyond the scope of this document (see **[5]** for more information).

Note – Documents used to provision the reference configuration are listed in **Section 11**. Specific references to these documents are indicated in the following sections by the notation **[x]**, where x is the document reference number.

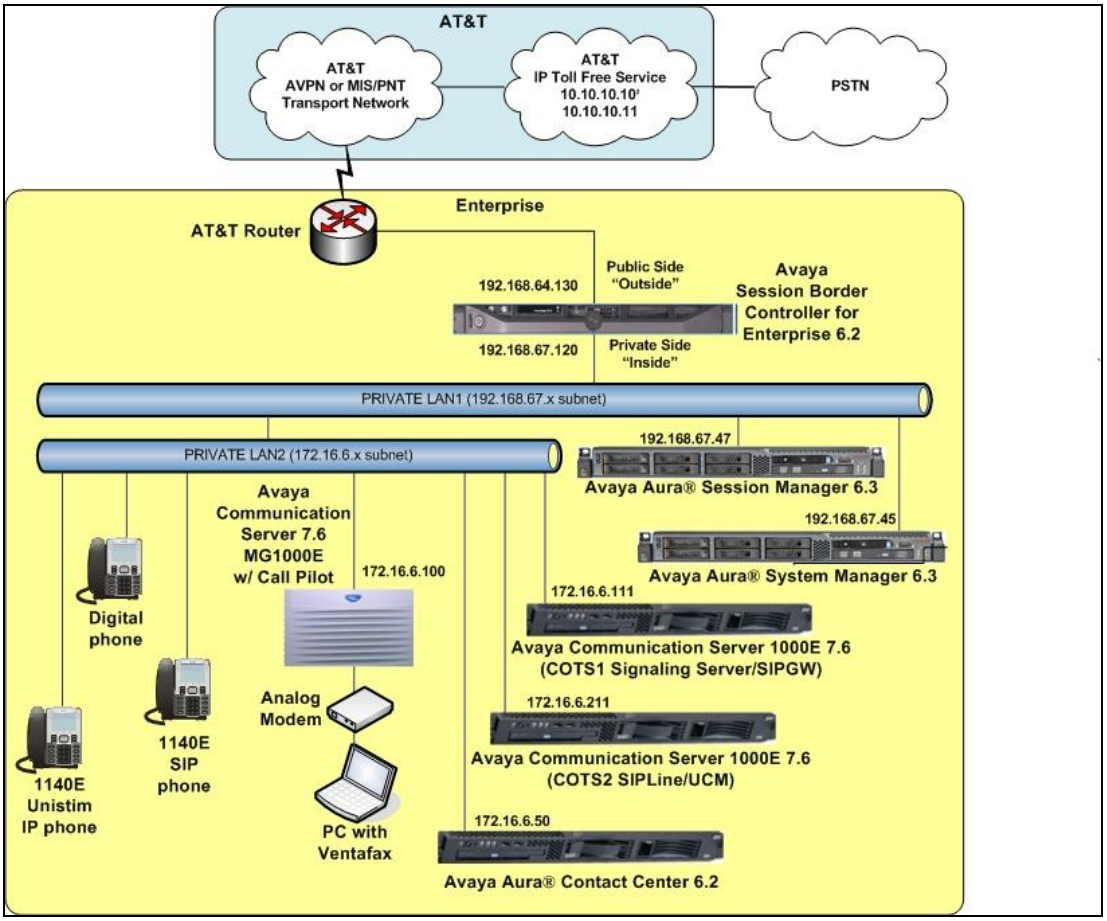


Figure 1: Avaya Interoperability Reference Configuration

[2] See the note on the next page regarding the IP addresses.

3.1. Illustrative Configuration Information

The specific values listed in **Table 1** below and in subsequent sections are used in the reference configuration described in these Application Notes, and are for illustrative purposes only. Customers must obtain and use the specific values for their own configurations.

Note – The AT&T IP Toll Free service Border Element IP addresses and DID/DNIS digits are shown in this document as examples. AT&T Customer Care will provide the actual IP addresses and DID/DNIS digits as part of the AT&T IP Toll Free provisioning process.

Component	Illustrative Value in these Application Notes
CS1000E	
COTS1 SIP Signaling Server IP Address (TLAN)	172.16.6.111
COTS2 SIP Line IP Address (TLAN)	172.16.6.211
MGC Media (DSP) IP Address (TLAN)	172.16.6.100
CS1000E extensions	40xx
Avaya Call Pilot®	
Call Pilot Application	172.16.6.12
Call Pilot Mailboxes	4xxx
Avaya Aura® Contact Center	
	172.16.6.50
Avaya Aura® System Manager	
	192.168.67.45
Avaya Aura® Session Manager	
	192.168.67.47
Avaya SBCE	
IP Address of "Outside" (Public) Interface (connected to AT&T Access Router/IP Toll Free Service)	192.168.64.130
IP Address of "Inside" (Private) Interface (connected to Session Manager)	192.168.67.120
AT&T IP Toll Free Service	
Border Element IP Addresses (Primary & Secondary)	10.10.10.10, 10.10.10.11*

Table 1: Illustrative Network Values Used in these Application Notes

***NOTE** – The Avaya SBCE Outside interface communicates with AT&T IP Toll Free Border Elements (BEs). For security reasons, the IP addresses of the BEs are not included in this document. However as a placeholder in the following sections, the IP addresses of **10.10.10.10** and **10.10.10.11** are used to represent the AT&T IP Toll Free BE IP addresses where required.

3.2. Call Flows

To understand how inbound AT&T IP Toll Free service calls are processed by Session Manager and CS1000E, two general call flows are described in this section.

3.2.1. Inbound Call to the Avaya CS1000E

The first call scenario illustrated in **Figure 2** is an inbound AT&T IP Toll Free service call that arrives on Session Manager and is subsequently routed to CS1000E.

1. A PSTN telephone originates a call to an AT&T IP Toll Free service number.
2. The PSTN routes the call to the AT&T IP Toll Free service network.
3. The AT&T IP Toll Free service routes the call to Avaya SBCE.
4. Avaya SBCE performs SIP Network Address Translation (NAT) and any necessary SIP header modifications, and routes the call to Session Manager.
5. Session Manager applies any additional SIP header adaptations and digit conversions, and based on configured Routing Policies, determines where the call should be routed next. In this case, Session Manager routes the call to CS1000E.
6. Depending on the called number, CS1000E routes the call to an agent or telephone.

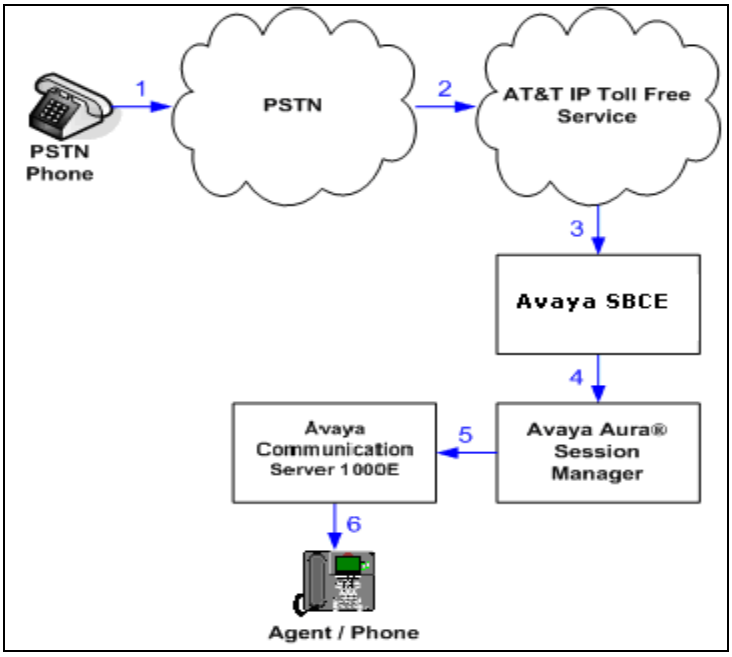


Figure 2: Inbound AT&T IP Toll Free Service Call to Agent / Telephone

3.2.2. Coverage to Voicemail

The call scenario illustrated in **Figure 3** is an inbound call that is covered to voicemail. In this scenario, the voicemail system is an Avaya Call Pilot® system connected to CS1000E.

1. Same as the first call scenario shown in **Section 3.2.1**.
2. The called CS1000E Agent/phone does not answer the call, and the call covers to the phone's voicemail. CS1000E forwards the call to Avaya Call Pilot®. Avaya Call Pilot® answers the call and connects the caller to the called phone's voice mailbox.

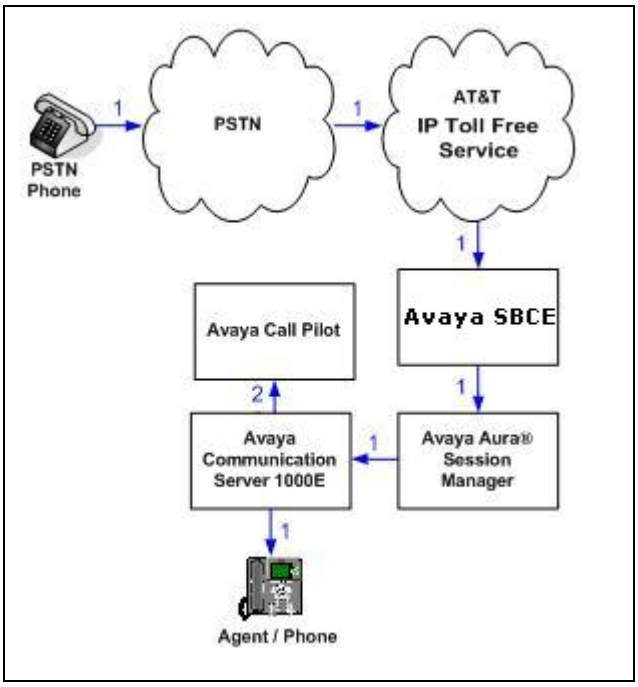


Figure 3: Inbound AT&T IP Toll Free Service Call - Coverage to Voicemail

4. Equipment and Software Validated

The following equipment and software was used for the reference configuration described in these Application Notes.

Equipment/Software	Release/Version
HP Proliant DL360 G7 server • System Platform • Avaya Aura® System Manager	• 6.3.0.0.18002 with patch 6.3.1.08002.0 • 6.3.5_r3501969 with patch 6.3.5_Patch1_r3502017
IBM 8800 server • Avaya Aura® Session Manager	• 6.3 SP5 (6.3.5.0.635005)
CS1000E Platform • Call Pilot	• Version 7.6 SP3 (SP_7.6_3) • cs1000-patchWeb-7.65.16.21-06.i386.000 • cs1000-linuxbase-7.65.16.21-08.i386.000 • cs1000-patchWeb-7.65.16.21-06.i386.000 • Deplists_CPM_X21_07_65P • CP 5.00.41
Dell R310 • Avaya Session Border Controller for Enterprise	• 6.2 Q48[2]
HP DL360 • Avaya Aura® Contact Center	• 6.2.205
Avaya 1140E Series IP Deskphones (UNIStim)	• 0625C8Q
Avaya 1140E Series IP Deskphones (SIP)	• SIP1140e04.03.12.00.bin
Avaya M3904 Series Digital Deskphones	-
Ventafax Home Version (Windows based Fax device)	• 6.1.59.144

Table 2: Equipment and Software Versions

[2] See Section 2.2.1, Item 6.

5. Configure Avaya Communication Server 1000E

This section describes the CS1000E configuration, focusing on the routing of calls to Session Manager over a SIP trunk. In the sample configuration, CS1000E Release 7.6 was deployed with Call Server applications running on a CPPM server platform with MGC, and utilizing servers running separate Signaling Server and SIP Gateway applications (COTS1), and SIPLINE and UCM applications (COTS2).

Session Manager Release 6.3 provides all the SIP Proxy Service (SPS) and Network Connect Services (NCS) functions previously provided by the Network Routing Service (NRS). As a result, the NRS application is not required to configure a SIP trunk between CS1000E and Session Manager Release 6.3. Therefore NRS was not included in the reference configuration.

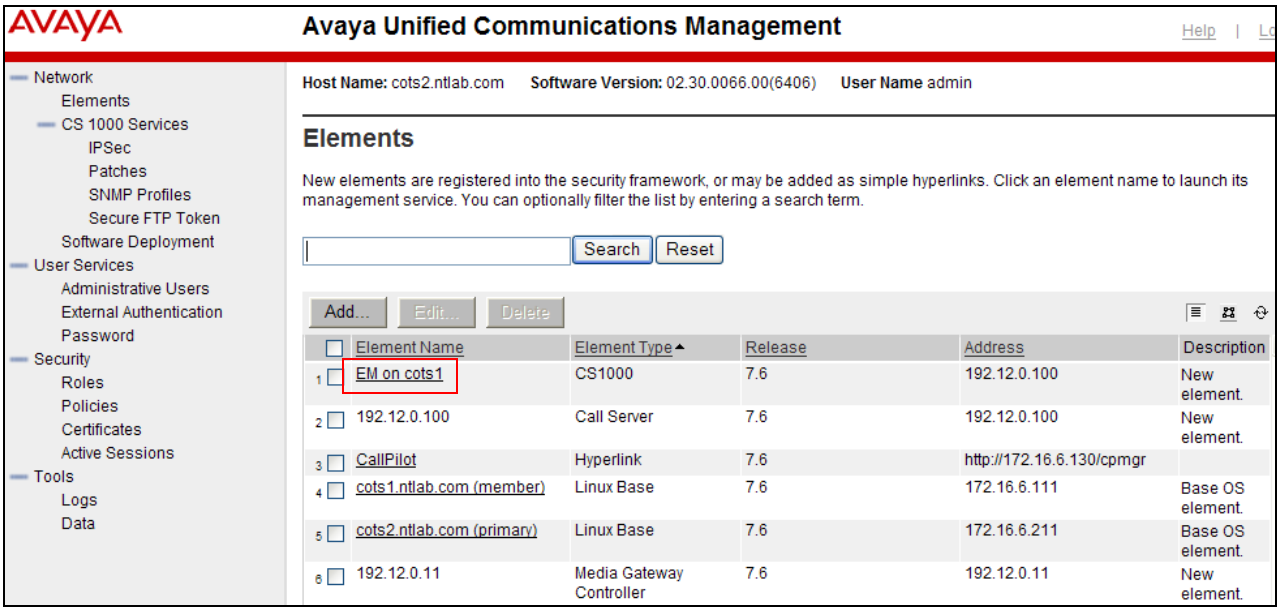
This section focuses on the SIP Trunking configurations for the CS1000E. Although sample screens are illustrated to document the overall configuration, it is assumed that the basic configuration of the Call Server and SIP Signaling Server applications has been completed, and that the CS1000E is configured to support analog, digital, UNIStim and SIP endpoints. For references on how to administer the CS1000E, see **Section 11.**

5.1. Logging In and Selecting the System Element

Step 1 - Unless otherwise noted, all CS1000E provisioning was performed via the Avaya Unified Communication Management (AUCM) web interface. The **AUCM** web interface may be launched directly via **https://<ip address>** where the relevant <ip address> in the sample configuration is 172.16.6.111. The following screen shows an abridged log in screen. Log in with appropriate credentials.

Note – Although not used in the reference configuration, System Manager may be configured as the Primary Security Server for the Avaya Unified Communications Management application and CS1000E is registered as a member of the System Manager Security framework. The Element Manager then may be accessed via the System Manager **UCM Services** link.

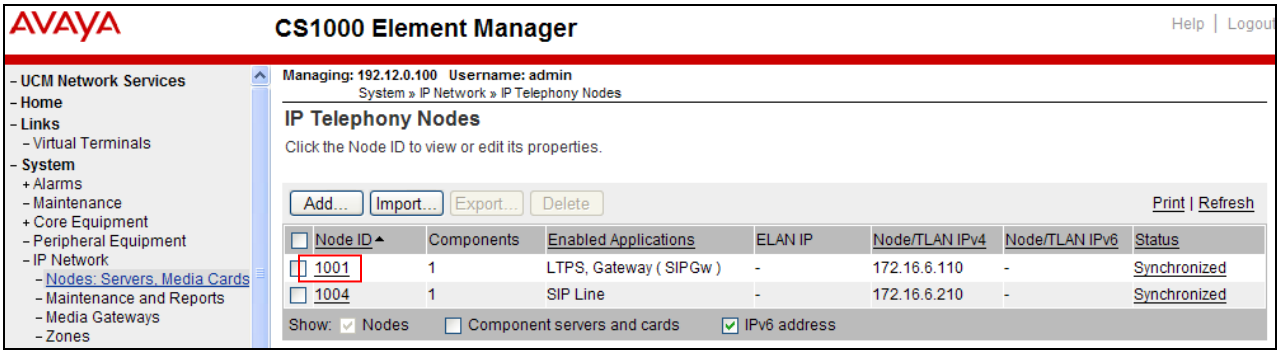
Step 2 - Click on the **Element Name** corresponding to **CS1000** in the **Element Type** column. In the sample screen below, the user would click on the **Element Name** "*EM on cots1*".



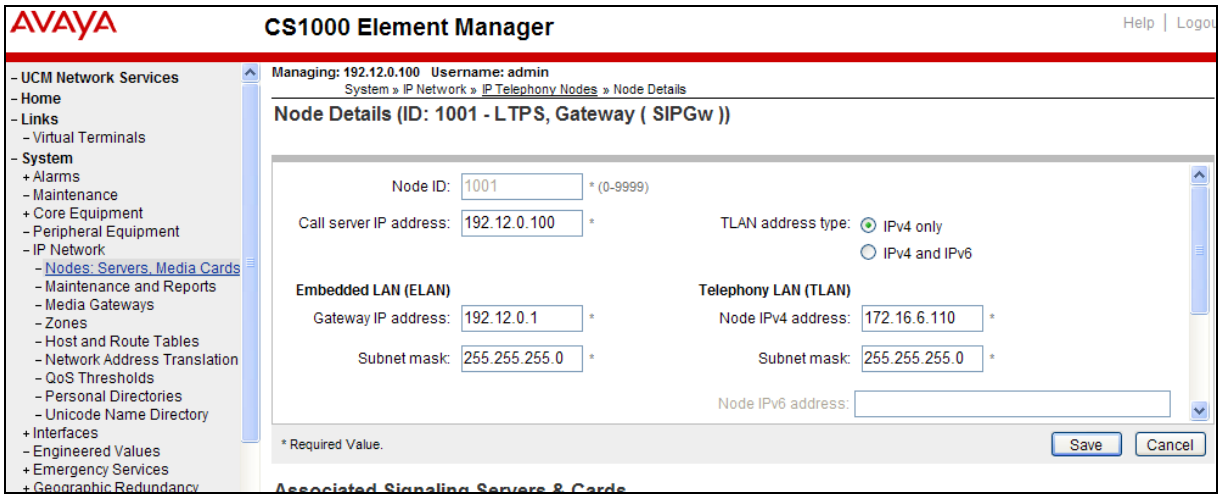
5.2. Administer Telephony Node

5.2.1. Node Information and IP Addresses

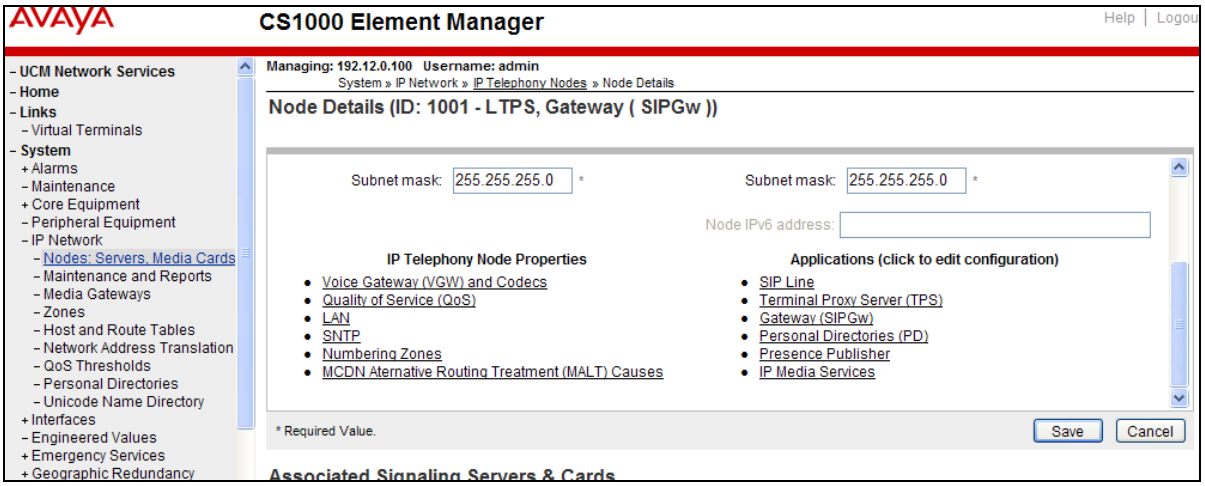
Expand **System → IP Network** on the left panel and select **Nodes: Servers, Media Cards**. The **IP Telephony Nodes** page is displayed as shown below. Click **<Node id>** in the **Node ID** column to view details of the node. In the sample configuration, node **1001** is selected.



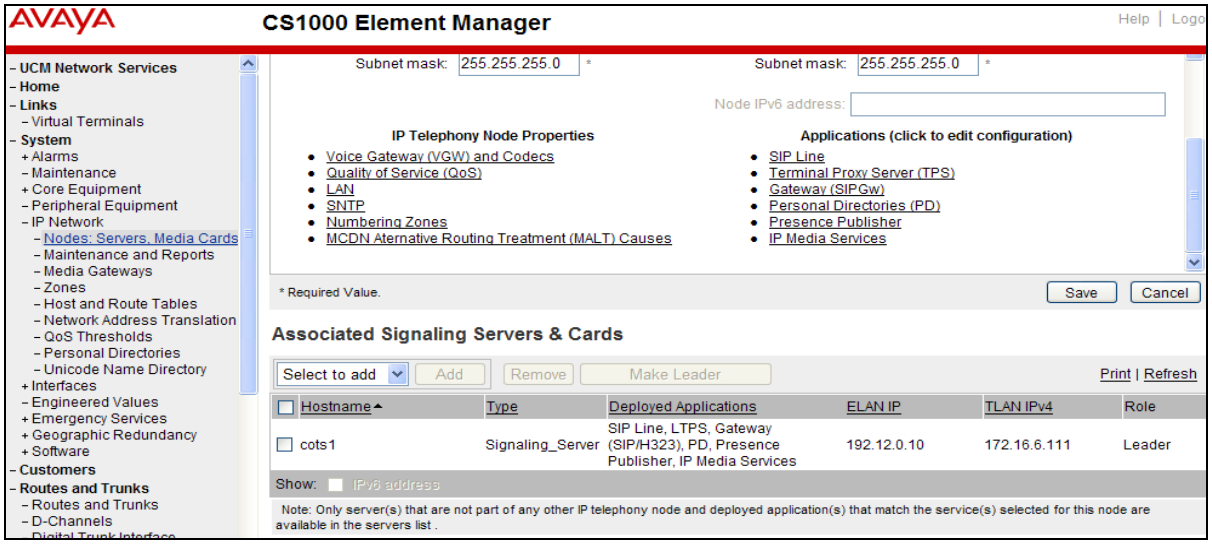
Under the **Node Details** heading at the top of the screen, make a note of the **TLAN Node IPV4 address.** In the sample screen below, the **Node IPV4 address** is 172.16.6.110**.** This IP address will be needed when configuring a Session Manager SIP Entity for CS1000E in **Section 6.4.2**.



Scrolling down the Node Details section, the various Node Properties and Applications may be selected.



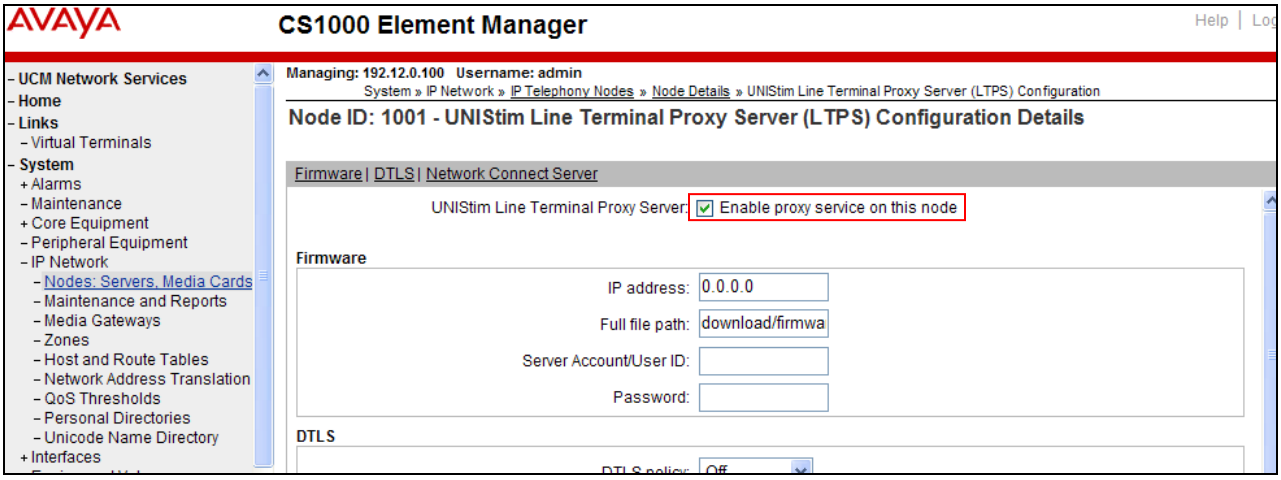
The **Associated Signaling Servers & Cards** information is displayed at the bottom of the screen.



5.2.2. Enable Terminal Proxy Server (TPS)

Continuing from **Section 5.2.1**, under the **Applications (click to edit configuration)** heading on the **Node Details** page, select the **Terminal Proxy Server** (**TPS**) application link.

Step 1 - Check the **UNIStim Line Terminal Proxy Server** checkbox to enable proxy service on this node.
Step 2 - Click on **Save** (not Shown).

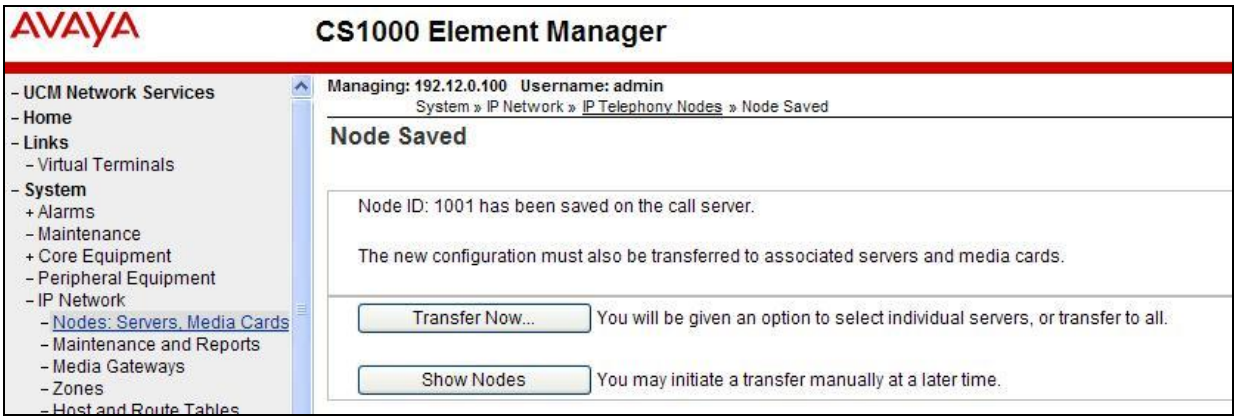


5.2.3. Synchronize Configuration

Step 1 - Scroll to the bottom of the page and click **Save**. This will return the interface to the **Node Details** screen.

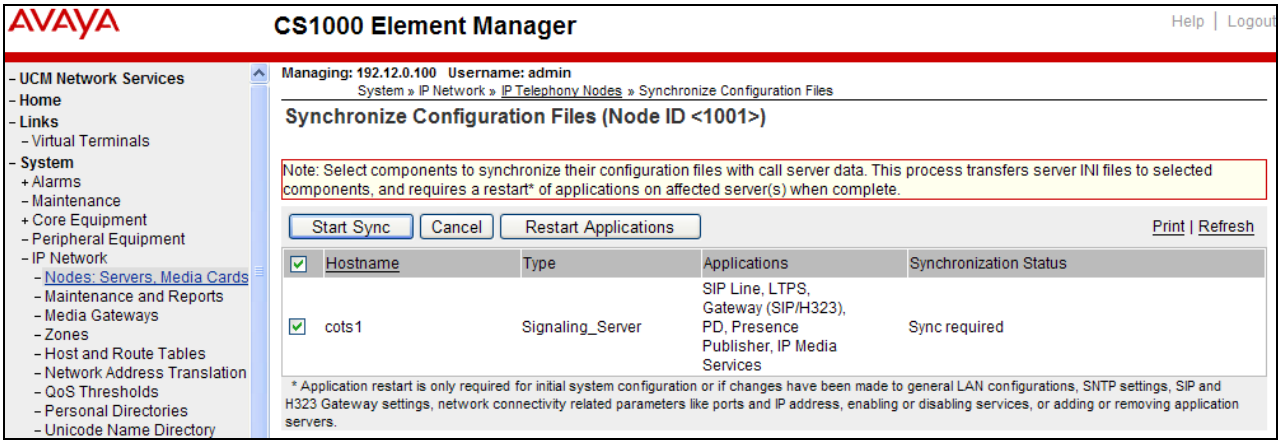
Step 2 - Click **Save** on the **Node Details** screen (not shown).

Step 3- Select **Transfer Now** on the **Node Saved** page as shown below.

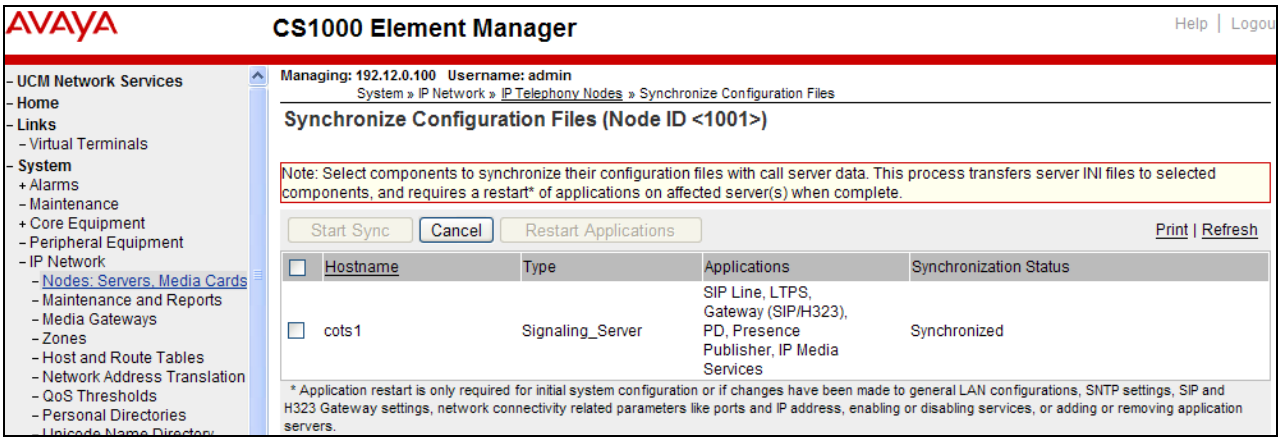


Once the transfer is complete, the **Synchronize Configuration Files (Node ID <id>)** page is displayed.

Step 4 - Select the appropriate Hostname (e.g., **cots1**) and click **Start Sync.**

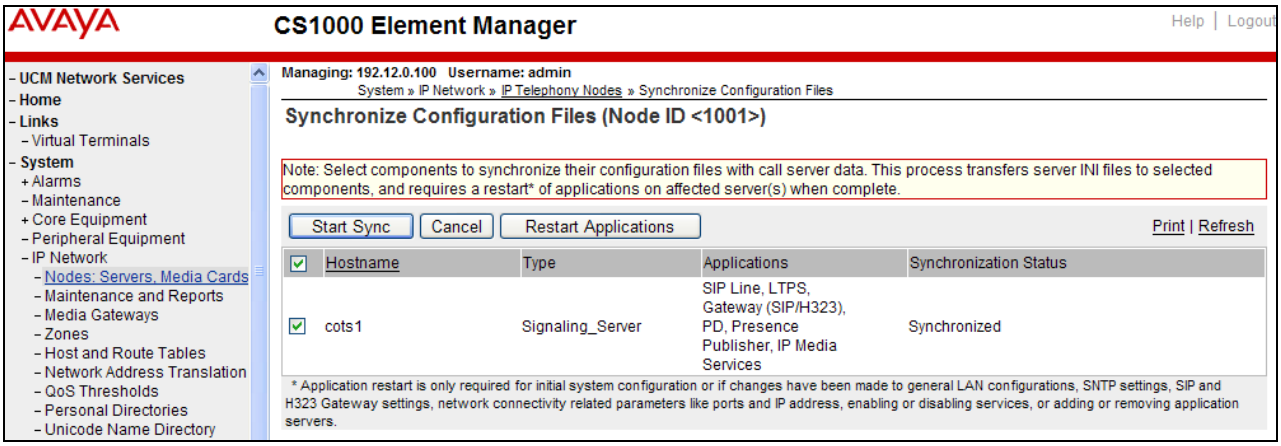


The Synchronization Status field will update from *Sync required*, to *Sync in progress*, to *Synchronized* as shown below



Step 5 - After synchronization completes, click on the **Refresh** button in the right hand corner, Select the appropriate Hostname (e.g., cots1), and click **Restart Applications**.

NOTE - When the applications restart, the phones will also reset.



5.3. Voice Codecs

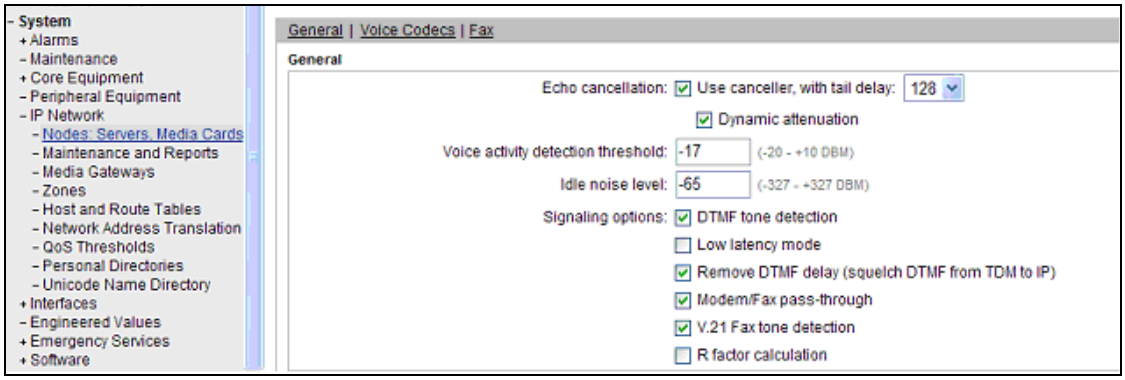
The following section describes how to set codec preferences as well as setting Packet Interval (PTIME) values. Note that the CS1000E always specifies G.711mu-law regardless of the additional selected codes. Codecs are defined in the **IP Telephony Node** for IP (e.g., UNIStim) phones, and the **Media Gateway** (for analog and digital phones).

5.3.1. IP Telephony Node Codec Configuration

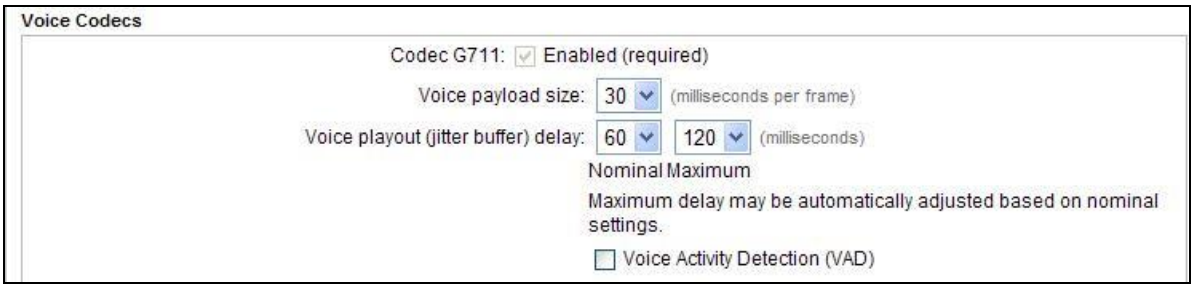
Step 1 – As shown in **Section 5.2**, expand **System → IP Network**, select **Node, Server, Media Cards**, and select node **1001**.

Step 2 – Scroll down the upper half of the form and under the **IP Telephony Node Properties** heading, select **Voice Gateway (VGW) and Codecs** (not shown).

The following screen shows the **General** parameters used in the sample configuration.



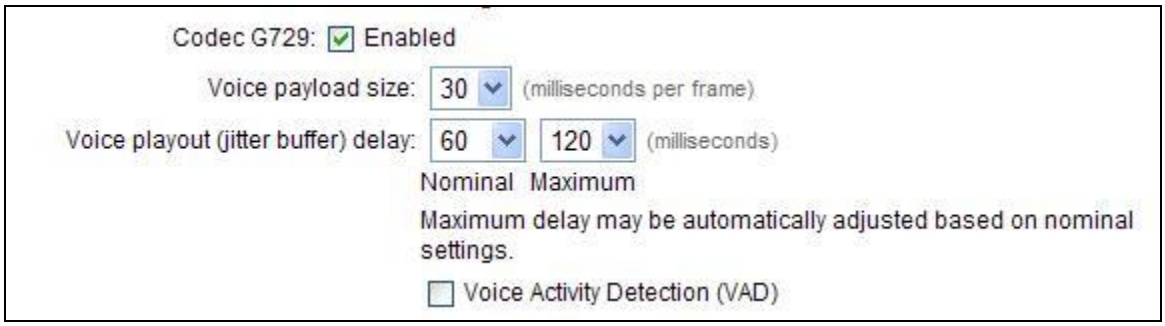
Step 3 - Use the scroll bar on the right to find the area with heading **Voice Codecs**. Set the **Voice payload size** to **30**. Note that **Codec G.711** is enabled by default.



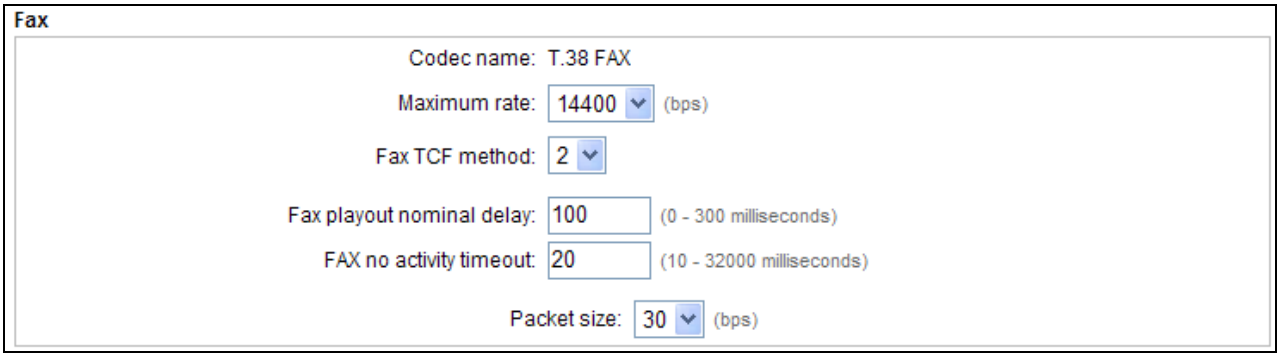
Step 4 – Scroll down to the G729 codec section and check the selection box. Set the **Voice payload size** to **30.**

Note – Although not shown, annexB=yes may be enabled by selecting the **VAD** (Voice Activity Detection) box. However, if enabled here, it should also be enabled in **Section 5.3.2**.

Note - On this screen, and the screen shown in **Step 3**, the Voice Payload size is set to 30ms. This is to meet the AT&T IP Toll Free recommendation. However an issue was found where the CS1000E would still send a payload size of 20ms (see **Section 2.2.1**, **Item 1**).



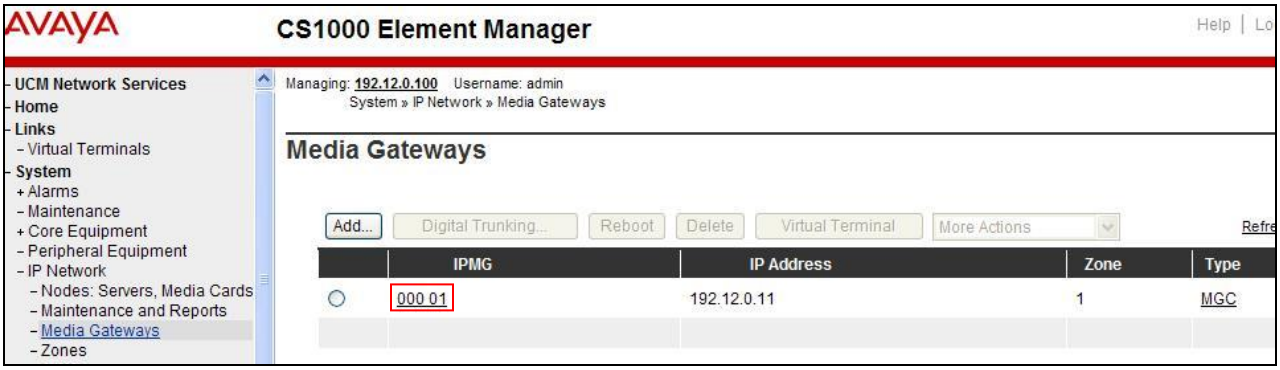
Step 5 - Scrolling further down, note that T.38 fax is enabled by default. Verify the **Maximum Rate** is set to **14400**.



Step 6 – Click on **Save** and then follow **Steps 1** through **5** in **Section 5.2.3** to synchronize the configuration.

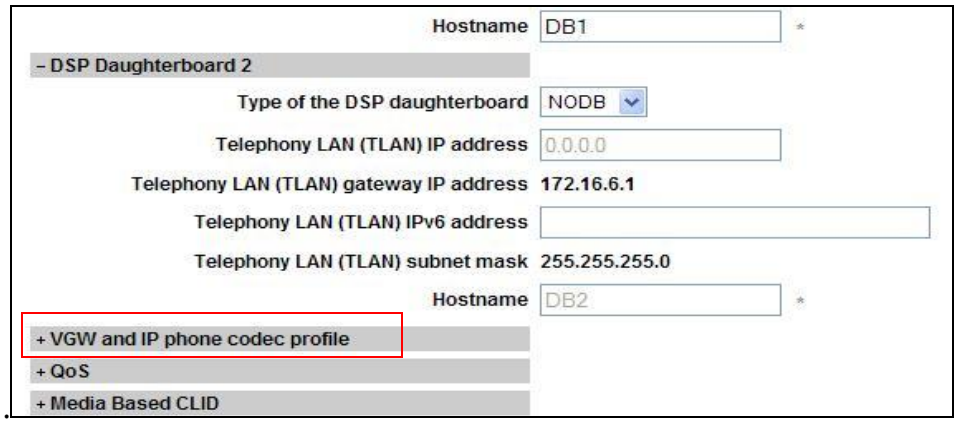
5.3.2. Media Gateway Codec Configuration

Step 1 - Expand **System → IP Network** on the left panel and select **Media Gateways**. Click on the IPMG ID (e.g., **000 01**).

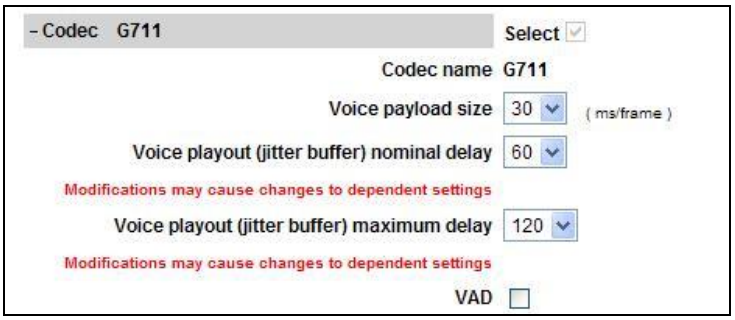


This will open the **Property Configuration** screen (not shown). Click on **Next** (not shown). This will open the **Media Gateway Controller (MGC) Configuration** screen.

Step 2 - Scroll down and click on **VGW and IP phone codec profile**.

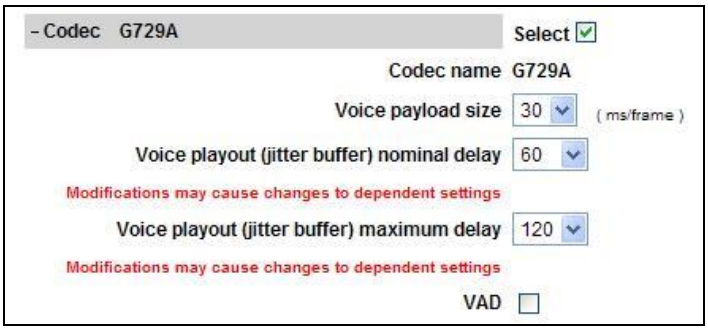


Step 3 - The **VGW and IP phone codec profile** section will expand. Scroll down, click on and expand the **Codec G711** field. Note that the "Select" box is checked by default. Set the **Voice payload size** (PTIME) to **30** (see the note in **Section 5.3.1** regarding payload size).

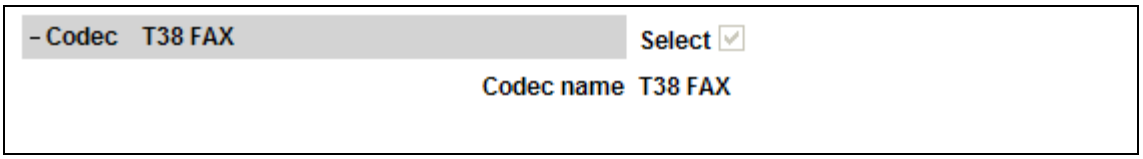


Step 4 – Scroll down, click on and expand the **Codec G729A** field. Check the selection box and set the **Voice payload size** (PTIME) to **30** (see the note in **Section 5.3.1** regarding payload size).

Note – Although not shown, annexB=yes may be enabled by selecting the **VAD** (Voice Activity Detection) box. However, if enabled here, it should also be enabled in **Section 5.3.1**.

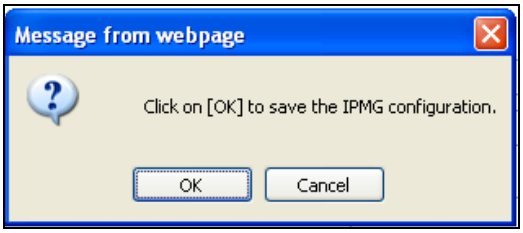


Step 5 – Scroll down and click on **Codec T.38 FAX**. Note that T.38 is enabled by default.

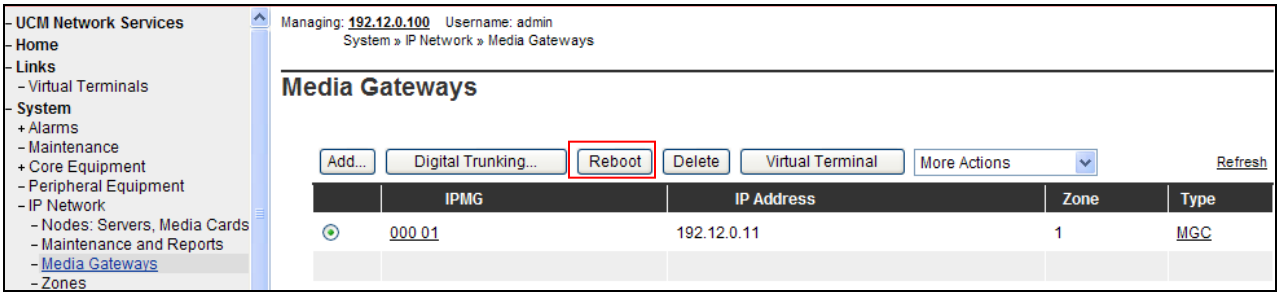


Step 6 – If changes are made to any of these settings, click on **Save** (not shown).

Step 7 – A dialog box will open. Click on **Ok**.



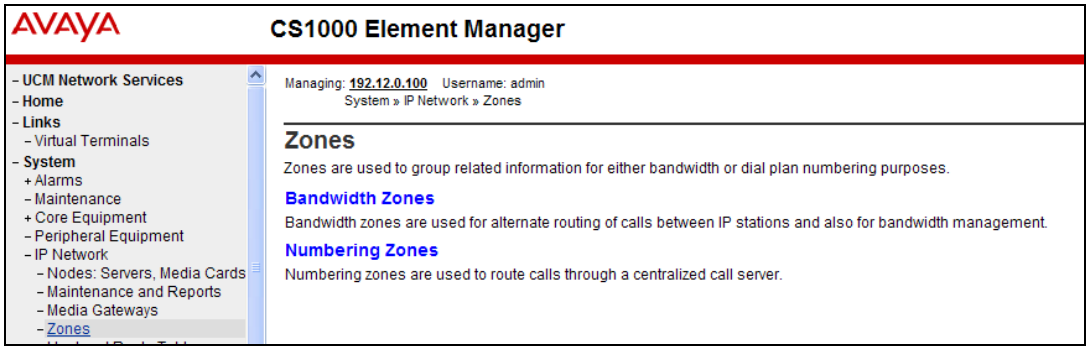
Step 8 –Select the Media Gateway ID (e.g., 000 01), and click on the **Reboot** button. The Media Gateway will reboot and deploy the new configuration.



5.4. Zones and Bandwidth Management

Zone configuration can be used to control codec selection and for bandwidth management.

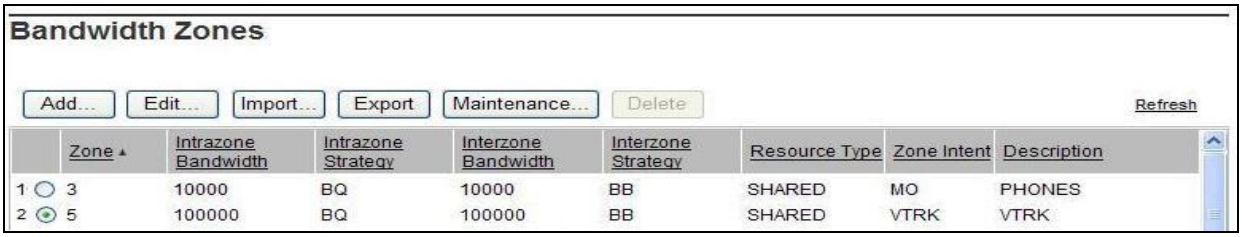
Step 1 - Expand **System → IP Network** and select **Zones** as shown below.



Step 2 - Select **Bandwidth Zones**. In the reference configuration, two zones are configured as shown below. **Zone 3** is for the IP telephones and **Zone 5** is for the SIP trunk. Additional zones may be added by selecting the **Add** button.

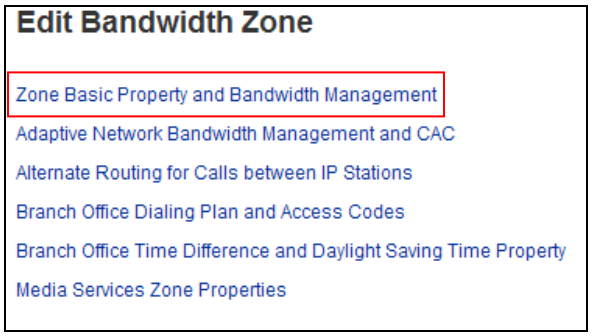
5.4.1. Zone 5 – SIP Trunk

Step 1 – Continuing from **Section 5.4**, **Step 2,** select the zone associated with the virtual trunk to Session Manager (e.g., zone **5**) and click **Edit** as shown below.

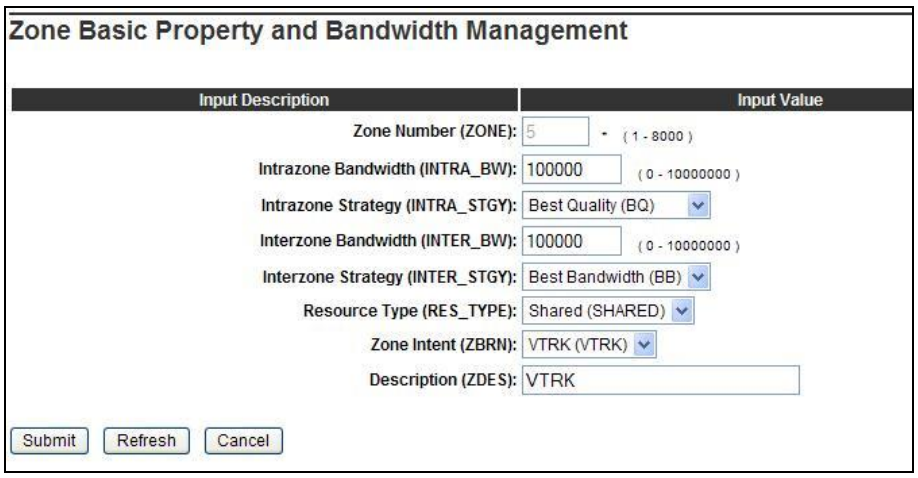


	Zone	Intrazone Bandwidth	Intrazone Strategy	Interzone Bandwidth	Interzone Strategy	Resource Type	Zone Intent	Description
1	3	10000	BQ	10000	BB	SHARED	MO	PHONES
2	5	100000	BQ	100000	BB	SHARED	VTRK	VTRK

Step 2 – Select **Zone Basic Property and Bandwidth Management** for Zone 5.

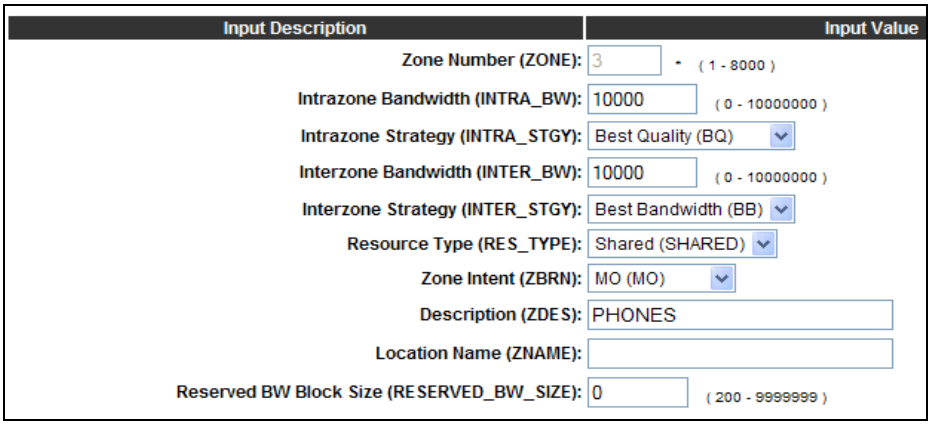


The following screen shows the **Zone 5** configuration. Note that the **Interzone Strategy** (access to the AT&T network) is set for "**Best Bandwidth (BB)**". This is so that codec G.729A is preferred over codec G.711mu-law for calls with the AT&T IP Toll Free service.



5.4.2. Zone 3 – IP Telephones

Following the steps in **Section 5.4.1**, these are the values used for **Zone 3** (IP Telephones), in the reference configuration.



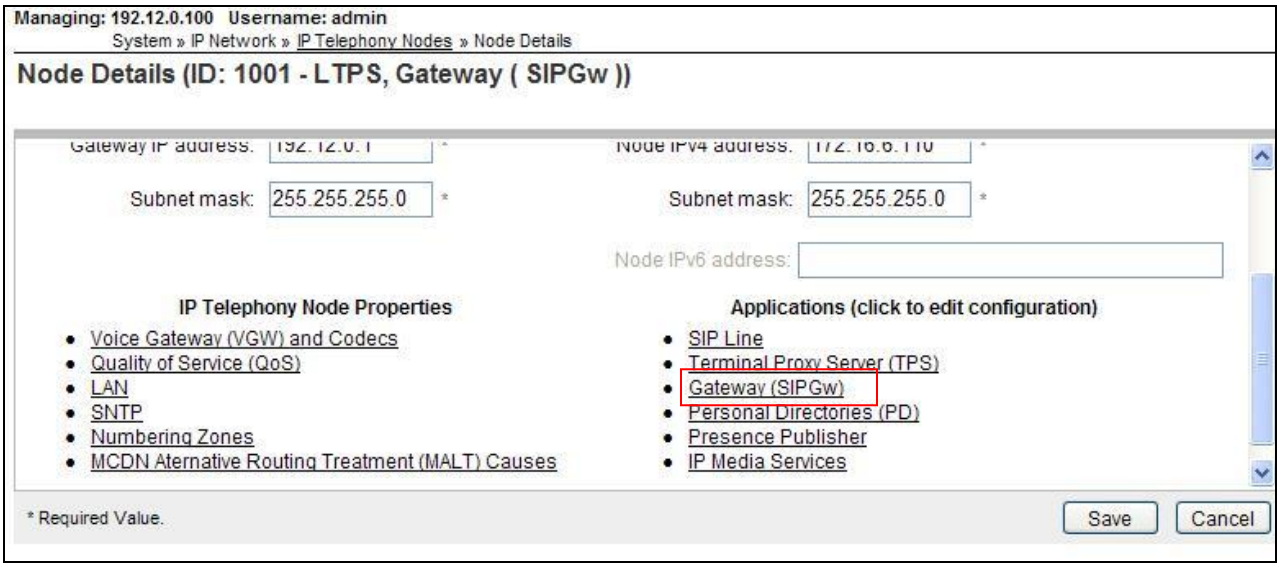
Input Description	Input Value
Zone Number (ZONE):	3 * (1 - 8000)
Intrazone Bandwidth (INTRA_BW):	10000 (0 - 10000000)
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	10000 (0 - 10000000)
Interzone Strategy (INTER_STGY):	Best Bandwidth (BB)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	MO (MO)
Description (ZDES):	PHONES
Location Name (ZNAME):	
Reserved BW Block Size (RESERVED_BW_SIZE):	0 (200 - 9999999)

5.5. SIP Trunk Gateway

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

5.5.1. Provision SIP Gateway

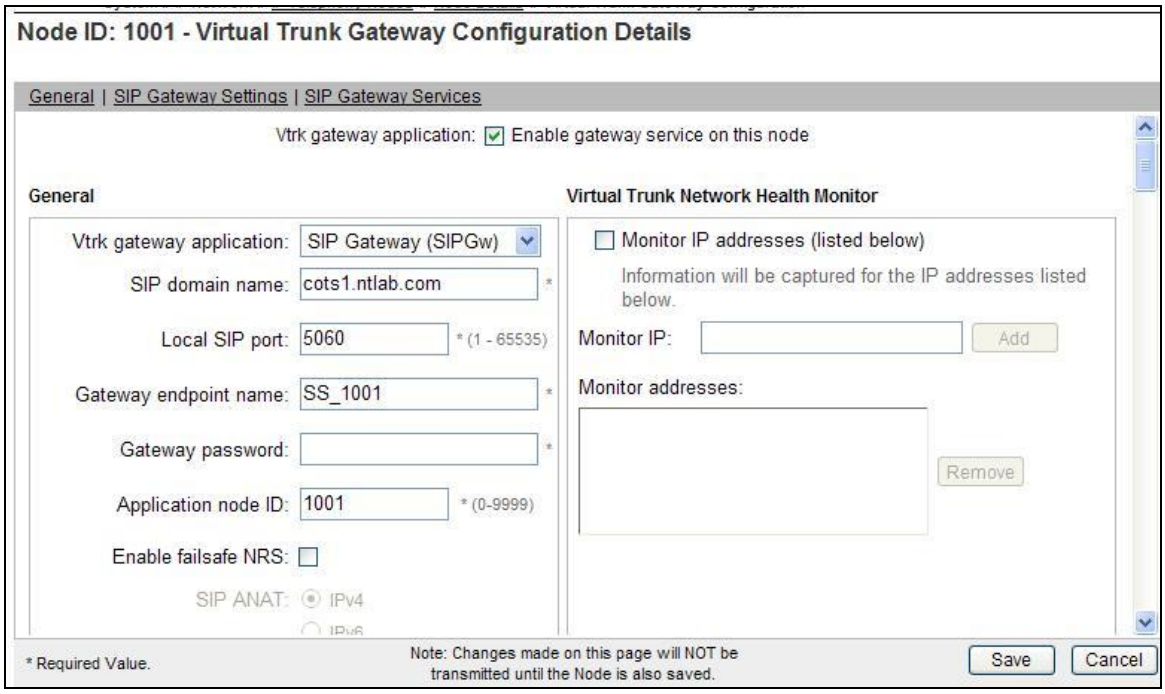
Step 1 – As shown in **Section 5.2.1**, expand **System → IP Network** on the left panel and select **Nodes: Servers, Media Cards**. Using the scroll bar on the right side of the screen, navigate to the **Applications** section on the screen and select the **Gateway (SIPGw)** link to view or edit the SIP Gateway configuration.



Step 2 - On the **Node ID: 1001 - Virtual Trunk Gateway Configuration Details** page, enter the following values and use default values for remaining fields.

- **SIP domain name:** Enter the appropriate SIP domain for the customer network. In the sample configuration, "**cots1.ntlab.com**" was used in the reference configuration (see **Section 6.1**).
- **Local SIP port:** Enter "**5060**"
- **Gateway endpoint name:** Enter descriptive name
- **Application node ID:** Enter "**<Node id>**". In the sample configuration, Node "**1001**" was used matching the node shown in **Section 5.2.1**.
- **VTrk gateway application**: select "**SIP Gateway (SIPGw)**".

The values defined for the sample configuration are shown below.

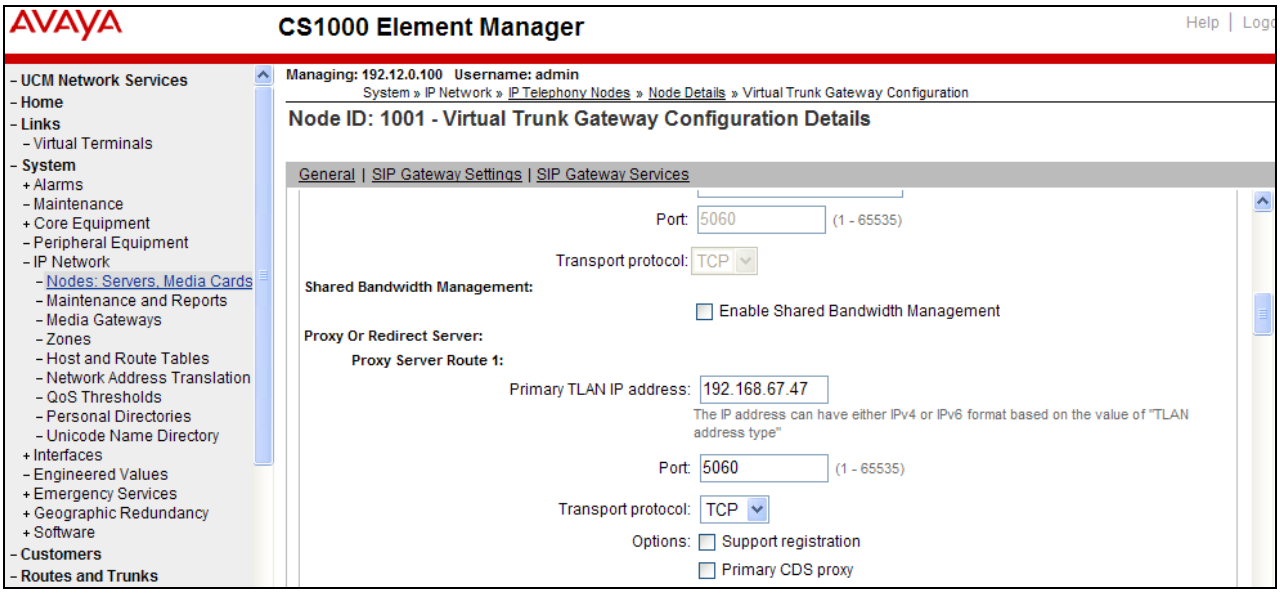


Step 3 - Scroll down to the section: **SIP Gateway Settings → Proxy or Redirect Server**.

Under **Proxy Server Route 1**, enter the following and use default values for remaining fields.

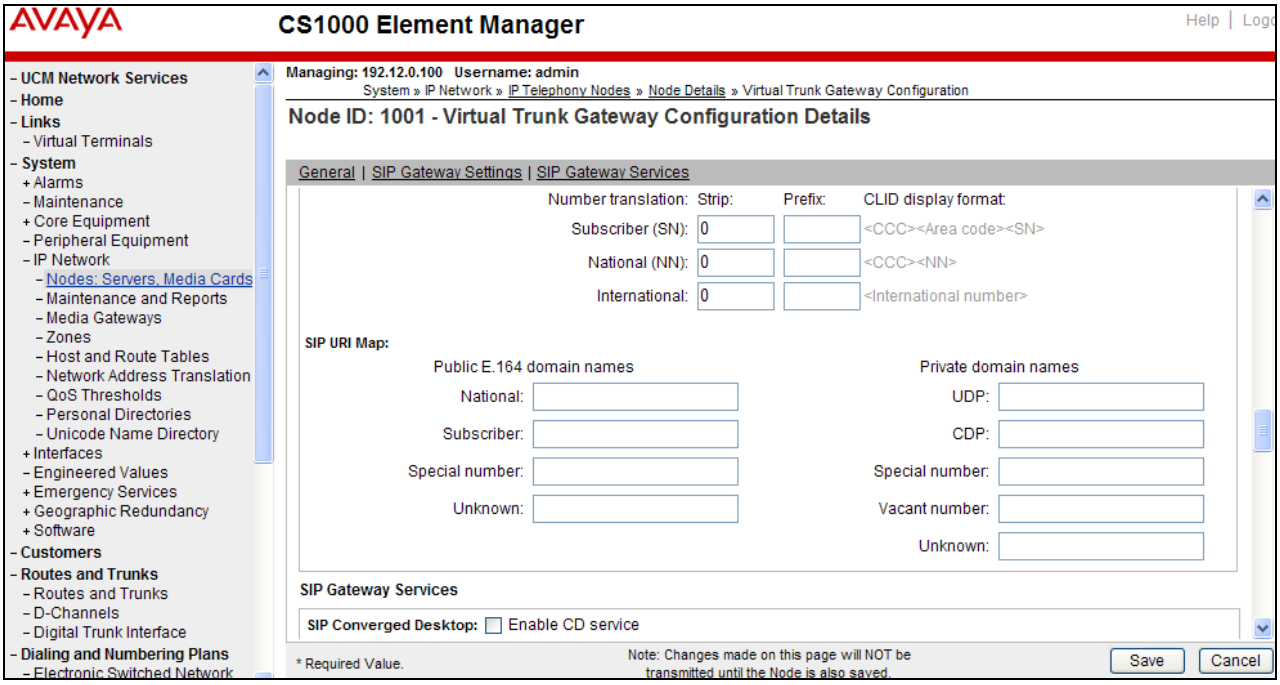
- **Primary TLAN IP address:** Enter the IP address of the Session Manager SIP signaling interface. In the sample configuration, "**192.168.67.47**" was used.
- **Port:** Enter "**5060**"
- **Transport protocol:** Select "**TCP**"

Note - The Secondary TLAN IP address was not used.



Step 4 - Scroll down and repeat these steps for the **Proxy Server Route 2** (not shown).

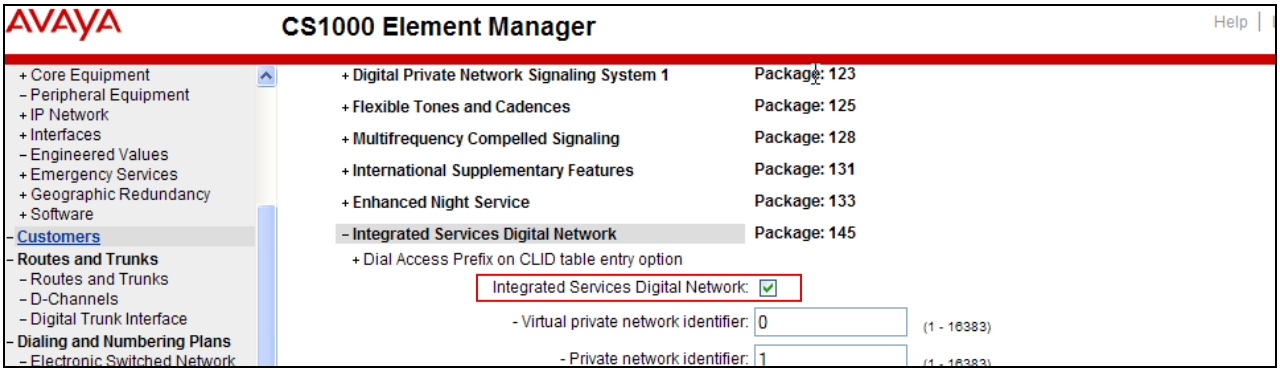
Step 5 - Scroll down to the **SIP URI Map** section. Under the **Public E.164 domain names** and **Private domain names** sections, leave the fields blank. Use the defaults for all other values.



Step 6 – Select **Save** and follow the synchronization steps shown in **Section 5.2.3**.

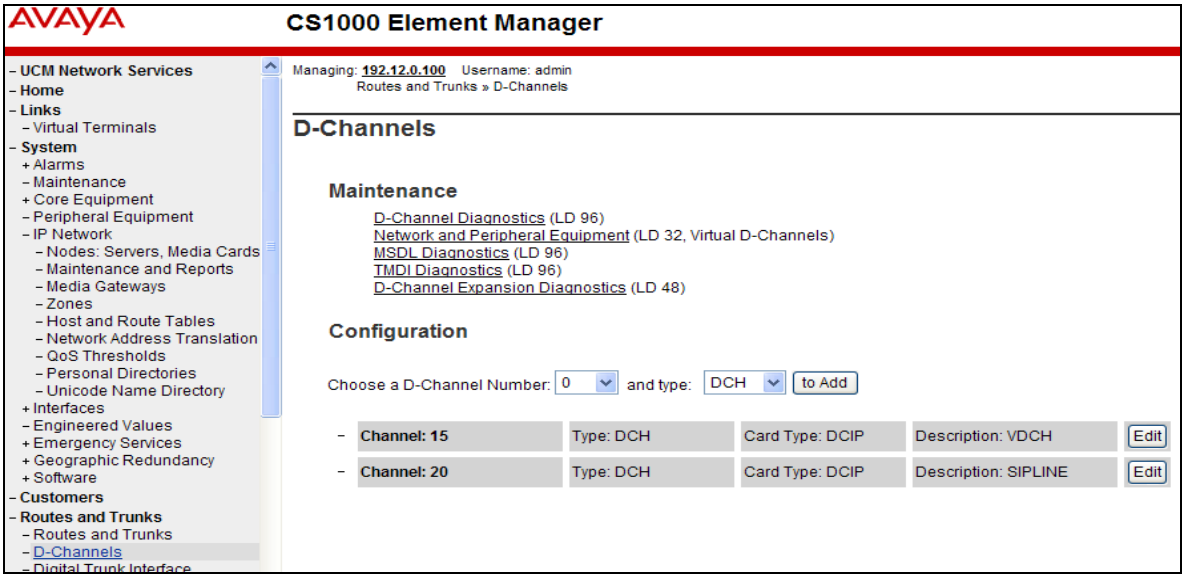
5.5.2. Integrated Services Digital Network (ISDN)

Step 1 - Select **Customers** in the left pane.
Step 2 - Click on the link associated with the appropriate customer, (e.g., **00**, not shown).The **Customer 00 Edit** page will appear (not shown).
Step 3 - Select the **Feature Packages** option from **Customer 00 Edit** page (not shown).
The screen is updated with a listing of available **Feature Packages**.
Step 4 - Select **Integrated Services Digital Network** to edit the parameters shown below. Check the **Integrated Services Digital Network** option, and retain the default values for all remaining fields. Scroll down to the bottom of the screen, and click on the **Save** button (not shown).

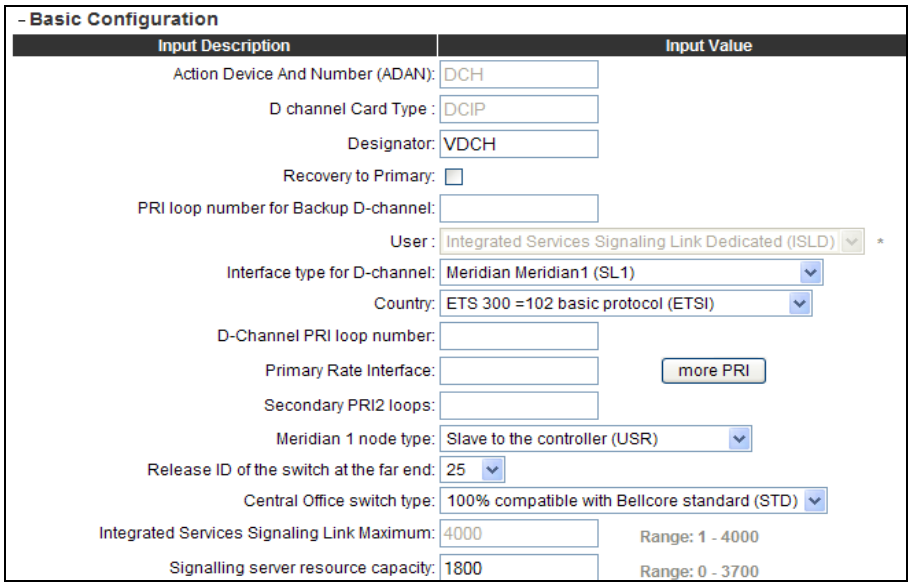


5.5.3. Virtual D-Channel Configuration

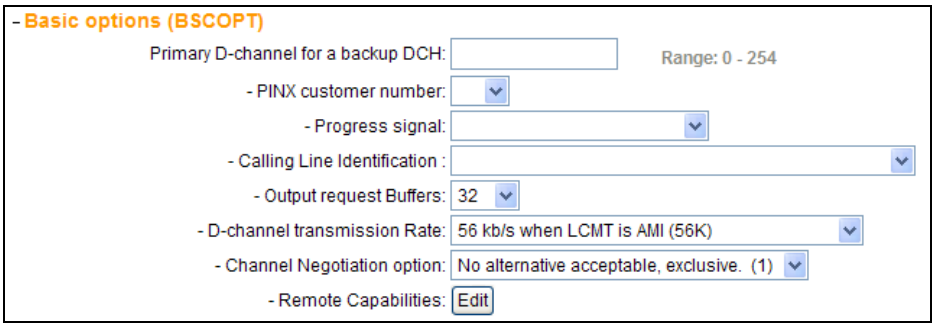
Step 1 - Expand **Routes and Trunks** on the left navigation panel and select **D-Channels**. In the sample configuration, **Channel 15** is associated with the Signaling Server. Channel 20 is associated with the SIP Line. Click on **Edit** to view/change settings. Click on the **To Add** button, to add additional D-Channels.



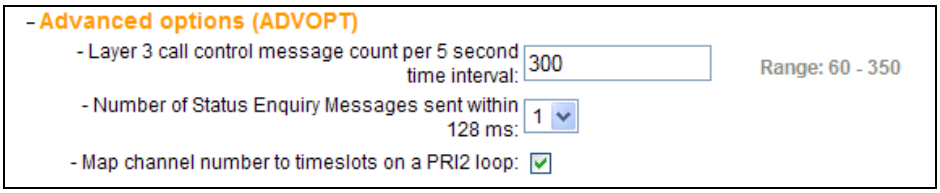
Step 2 – Click on **Edit** to display the associated D-Channel information used in the reference configuration for the Signaling Server (e.g., channel 15). The **D-Channels 100 Property Configuration** screen is displayed. In the **Basic Configuration** section, the following settings are used.



Step 3 – Scrolling down, in the **Basic Options (BSCOPT)** section, the following settings are used.



Step 4 – Scrolling down, in the **Advanced Options (ADVOPT)** section, the following settings are used.



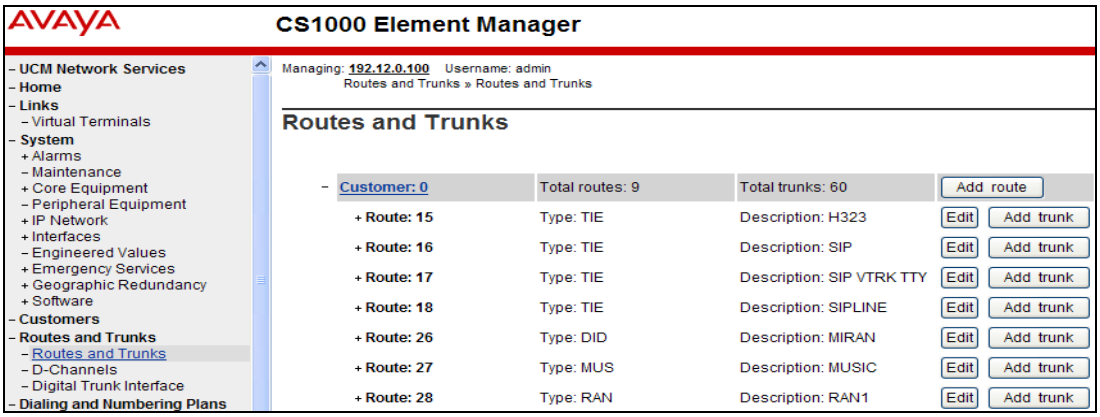
Step 5 – Click on **Submit** (not shown).

Step 6 – Repeat **Steps 1-5** to create the D-channel (e.g., **20**) for the SIP Line.

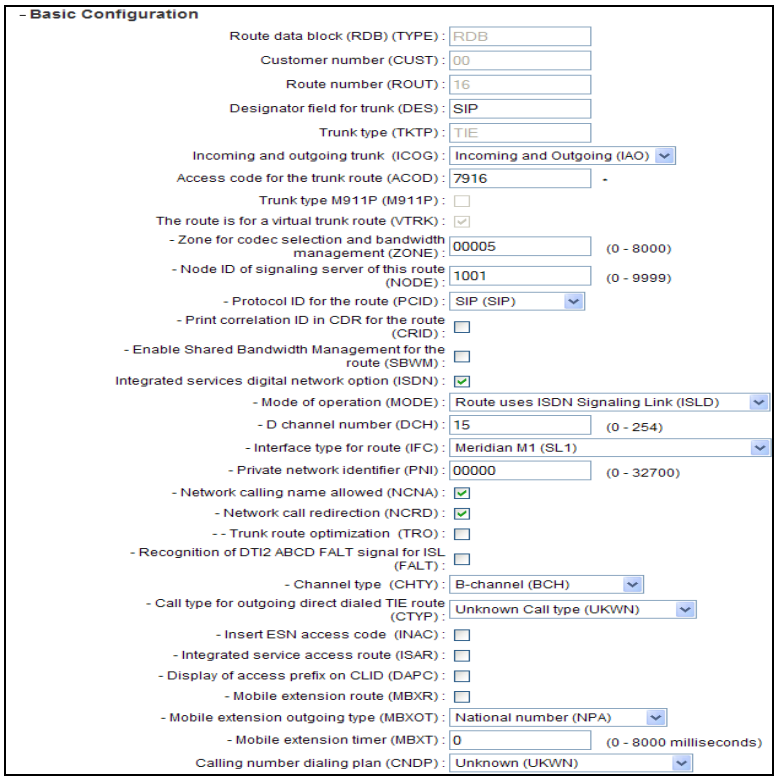
5.5.4. SIP Routes Configuration

Step 1 - Select **Routes and Trunks → Routes and Trunks** from the left pane to display the **Routes and Trunks** screen. In the reference configuration, **Customer 0** is used. Click on **Customer:0** to display defined routes, or click on **Add route**, to add additional routes.

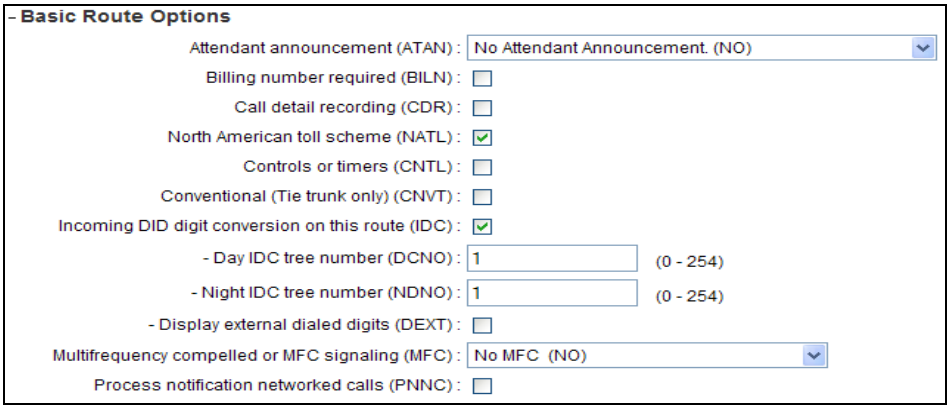
Step 2 – In the reference configuration, **Route 16** is used for SIP trunking. Click on the **Edit** button to display the Route 16 settings.



The following screen shows **Basic Configuration** settings for Route 16.

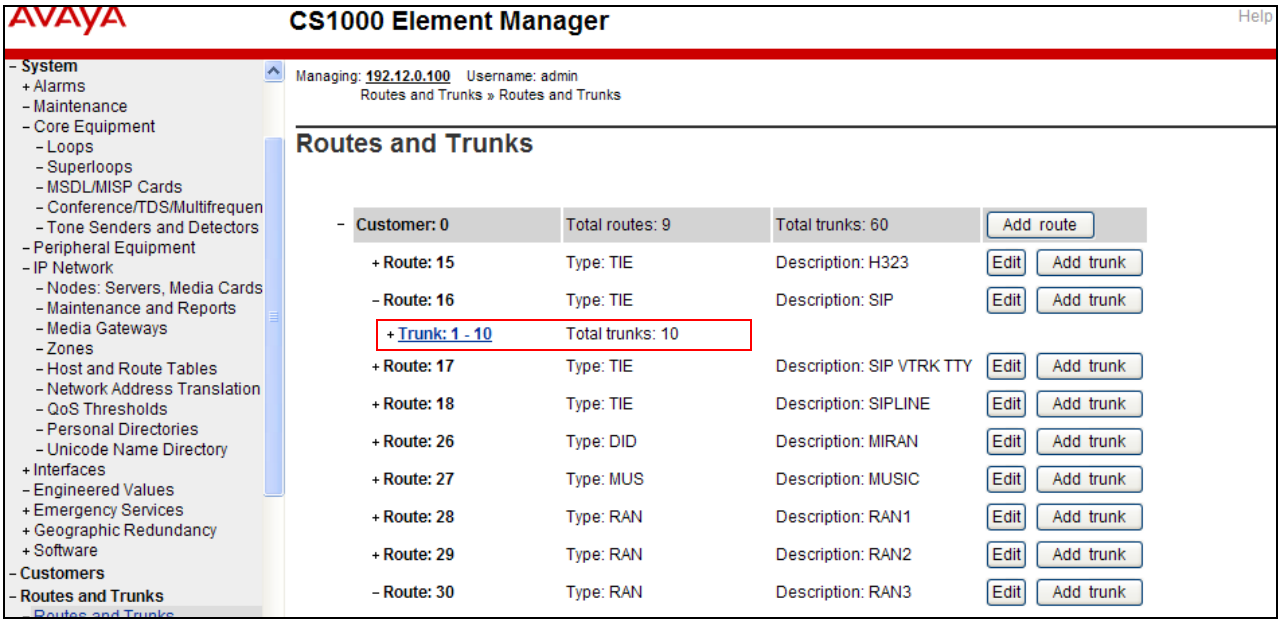


Step 3 – Scrolling down, click on **Basic Route Options**. The following settings are used in the reference configuration.

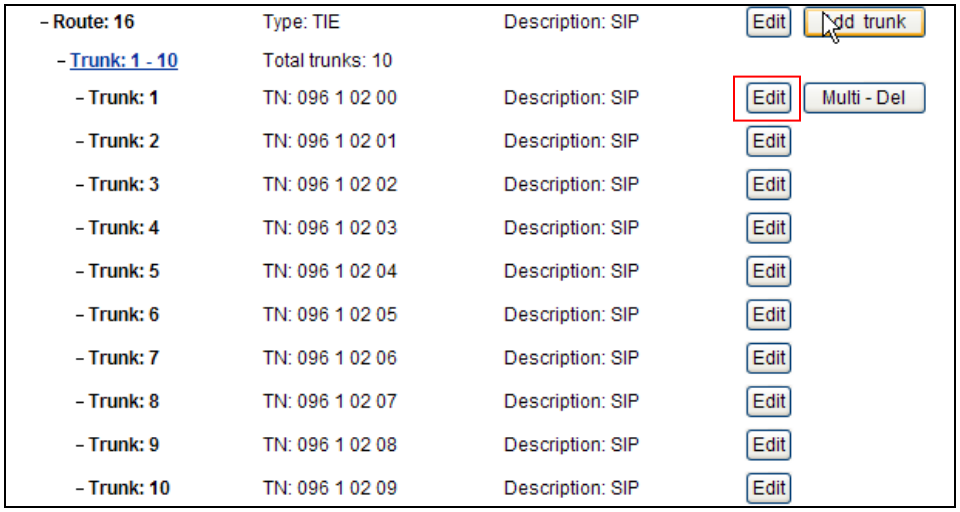


5.5.5. SIP Trunk Configuration to Session Manager

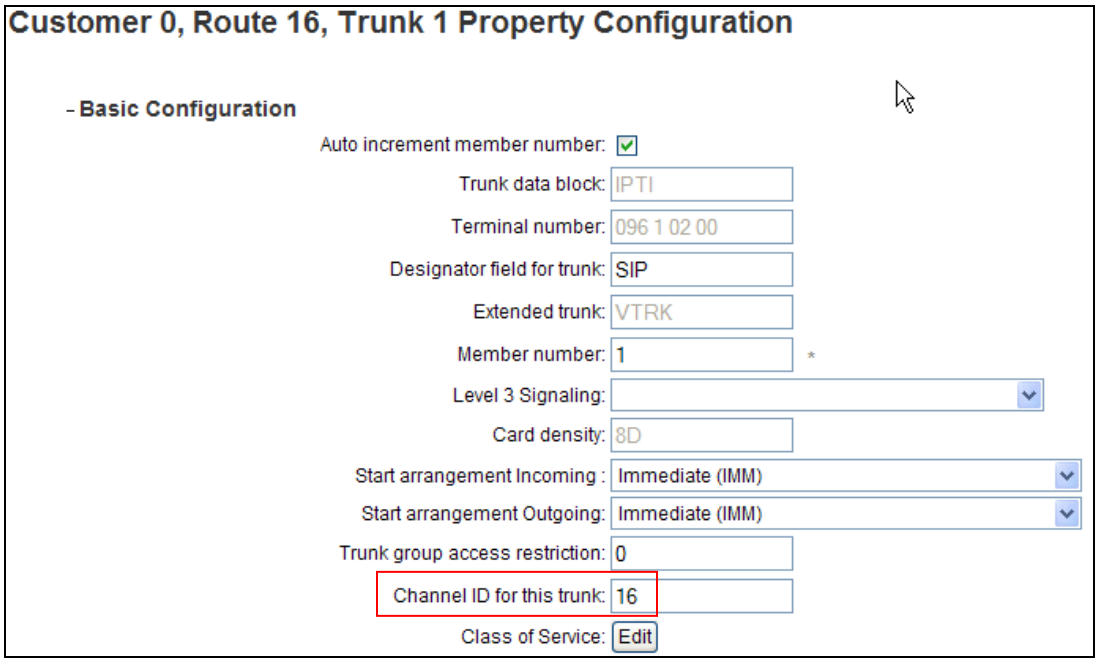
Step 1 - Select **Routes and Trunks → Routes and Trunks** on the left navigation panel and expand the **Customer 0**. Select **Route 16**, to display the 10 trunks used in the reference configuration (**Trunk:1 – 10**), or click **Add Trunk** to add additional trunks to the route.



Step 2 – Click on **Trunk:1-10** to display each trunk channel.

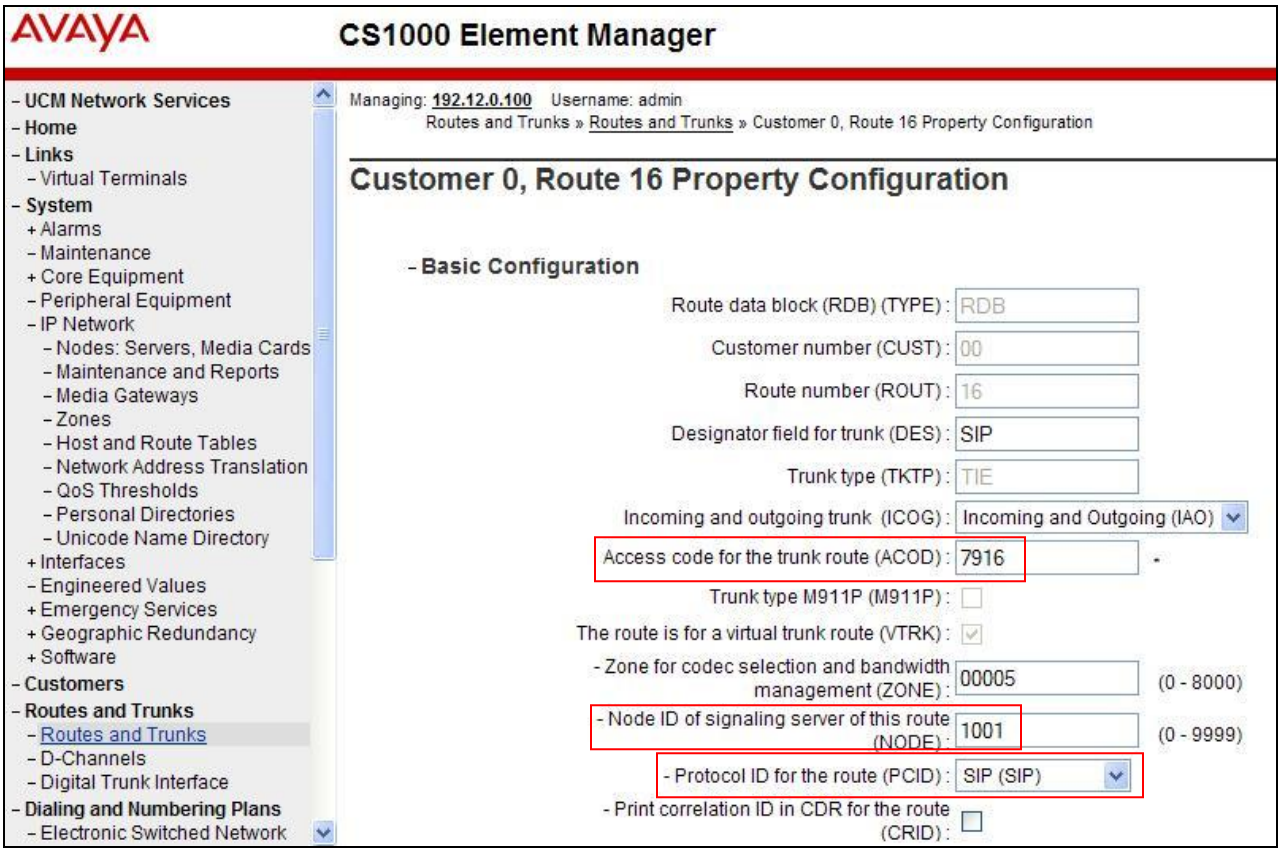


Step 3 – Click on the **Edit** button for **Trunk: 1**, to display the trunk configuration. In the reference configuration, Trunk 1 uses **Channel 16**. Therefore, each subsequent trunk allocated to this route will use channel 16+(n-1), where n is the trunk number. For example, Trunk 9 will use channel 24 (16+9-1 = 24).

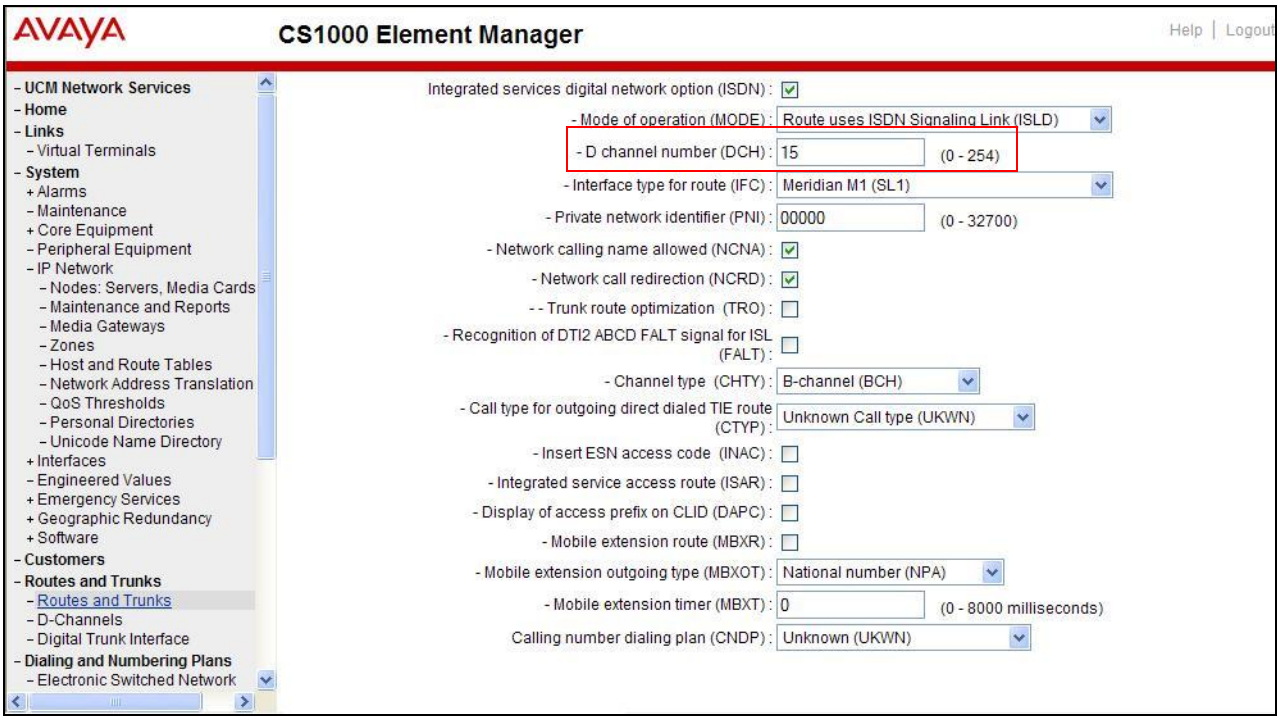


Step 4 – Going back to the screen shown in **Step 1**, select the **Edit** button next to **Route 16** to verify the configuration, as shown below. Verify "**SIP (SIP)"** has been selected for **Protocol ID for the route (PCID)** field and the **Node ID of signaling server of this route (NODE)** matches the node shown in **Section 5.2**. As can be observed in the **Incoming and outgoing trunk (ICOG)** parameter, incoming and outgoing calls are allowed. The **Access code for the trunk route (ACOD)** will in general not be dialed, but the number that appears in this field may be observed on CS1000E display phones if an incoming call on the trunk is anonymous or marked for privacy. The **Zone for codec selection and bandwidth management (ZONE)** parameter can be used to associate the route with a zone for configuration of the audio codec preferences sent via the Session Description Protocol (SDP) in SIP messaging.

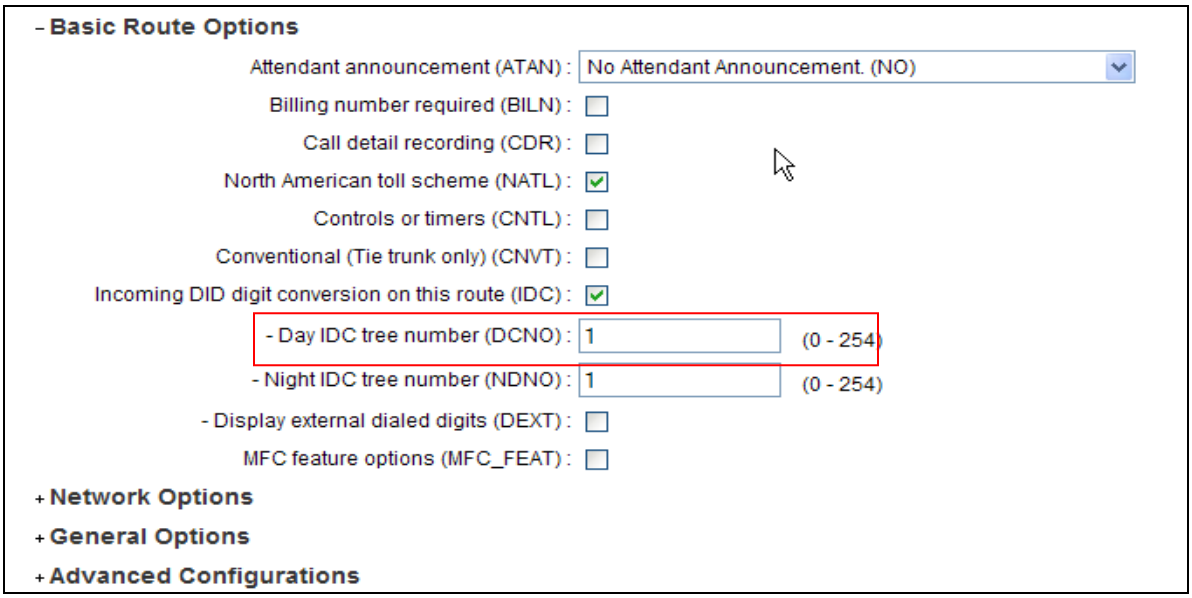
Note – Although the AT&T IP Toll Free service is an inbound only service, the trunks are configured for outbound as well, to facilitate SIP trunk calls to other destinations in the CPE via Session Manager (if needed).



Step 5 - Scrolling down, other parameters may be observed. The **D channel number (DCH)** field must match the D-Channel number shown in **Section 5.5.3** (e.g., **15**).



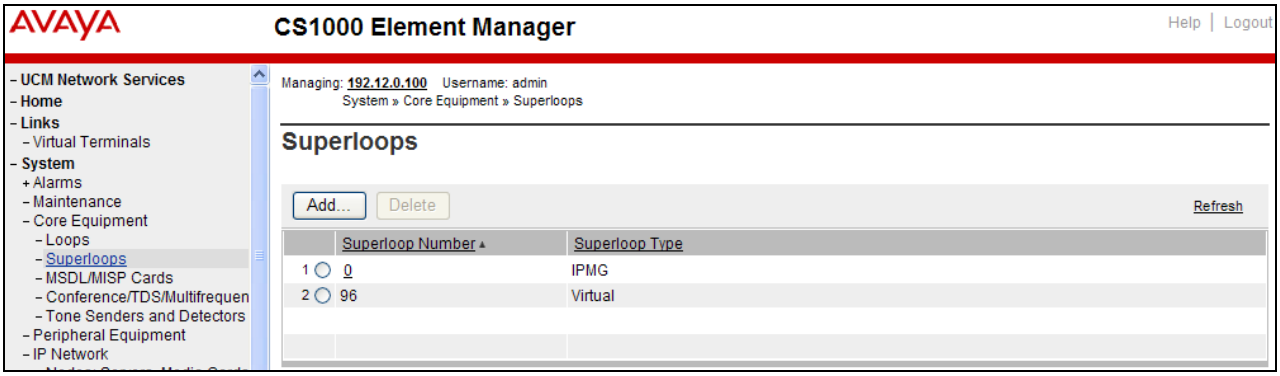
Step 6 - Scrolling down, open **Basic Route Options** and verify that the DCNO number specified (e.g., **1**), matches the **Digit Conversion Tree Number** specified in **Section 5.6, Step 3**.



Step 7 – After any changes or additions, click on **Submit** (not shown).

5.5.6. Administer Virtual Super-Loop

Select **System → Core Equipment → Superloops** from the left pane to display the **Superloops** screen. In the reference configuration, Superloops 0 and 96 are used.



5.6. Routing of Inbound Numbers to CS1000E

PSTN callers will dial AT&T IP Toll Free DID numbers to reach stations on CS1000E. The AT&T will then send associated DNIS digits in the Request-URI of the inbound SIP INVITEs. These DNIS digits are converted to their associated extensions by the CS1000E Incoming Digit Translation (IDT) table.

Note – The DNIS digits might not be the same as the dialed DID number[3].

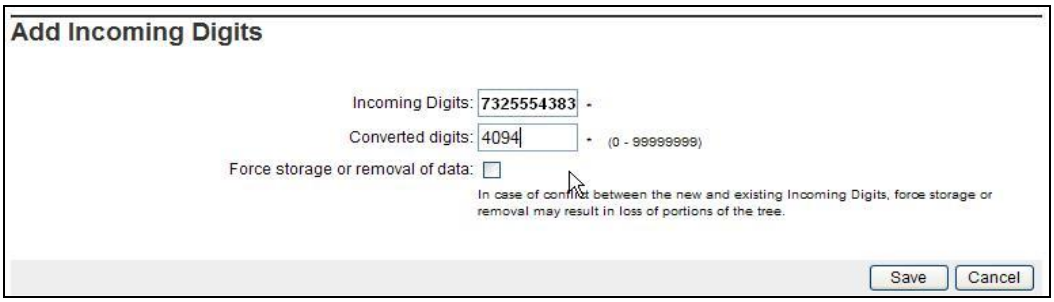
Step 1 – Navigate to **Dialing and Numbering Plans → Incoming Digit Translation**

Step 2 – Select the appropriate **Customer ID** (e.g., **00**) and click on **Edit IDC**.

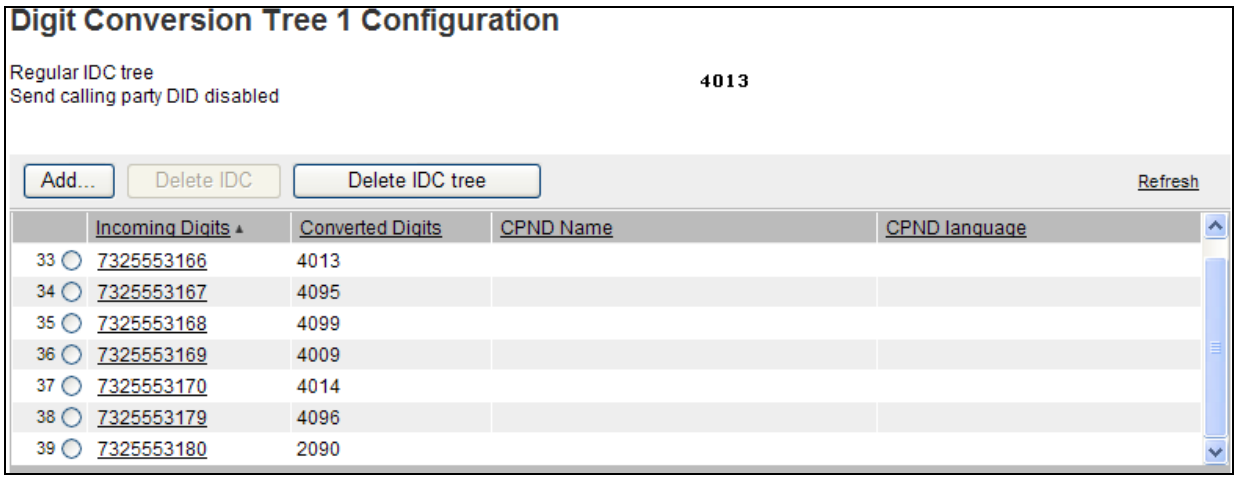
Step 3 – From the listed Digit Conversion Trees, select either **New DCNO** or edit **DCNO**. In the reference configuration, **Digit Conversion Tree Number: 1** was selected. Note that the **Digit Conversion Tree Number** selected must also be defined in the trunk provisioning shown in **Section 5.5.5**.

[3] See the issue described in Section 2.2.1, Item 5.

Step 4 – The IDC Tree form will open. Click on the **Add** button (not shown). In the **Incoming Digits** field, enter an AT&T IP Toll Free DNIS number sent in the R-URI (e.g., **7325554383**), and in the **Converted Digits** field, enter the associated CS1000E extension (e.g., **4094**).
Click on **Save**.



Step 5 – Repeat **Step 4** for all AT&T IP Toll Free DNIS numbers and their associated destination extensions. For example, a CS1000E Agent phone (e.g., 4014, see **Section 5.7**), an Agent skill queue (e.g., 4013, see **Section 5.7**), and the Call Pilot main access number (e.g., 2090, see **Section 5.10**).



5.7. CS1000E Agent Access Provisioning

This section is not intended to be prescriptive, but simply illustrates a sampling of defining Agent access on the CS1000E in the sample configuration. Inbound IP Toll Free numbers are mapped to the Agent extensions (or queues) as shown in **Section 5.6**.

The following Directory Numbers (DN) are defined:

- **2003** – This is the Positional DN. It is associated with the Terminal Number (TN) defined for an Agents phone (e.g., **96 0 1 17**).
- **4012** – This is the Auto Call Distribution (ACD) number for the agent queue. All agents share this queue. This number will appear on the Agent phone display.
- **4013** – This is the Control DN (CDN). It is used to define the connection between the CS1000E and the Avaya Aura® Contact Center (see **Section 5.7.3**).
- **4014** – This is the Agents Single Call Ringing (SCR) number. This is the Agents "local" extension independent of the Agent queue, and will also appear on the phone display. The Agent logs in with this number.

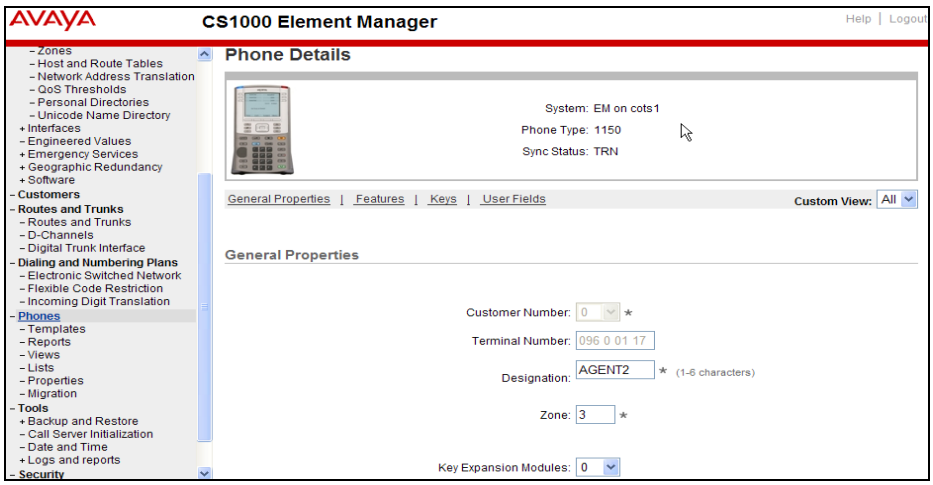
5.7.1. CS1000E IP Agent Phone

The following section shows information for an 1150E IP UNIStim Agent phone in the reference configuration defined via AUCM.

5.7.1.1 General Properties

Step 1 – Select **Phones** from the menu The **Search For Phones** screen will open (not shown).. In the **Criteria** field select **Prime DN** and enter a DN in the value field (e.g., **2003**). Click on **Search**.

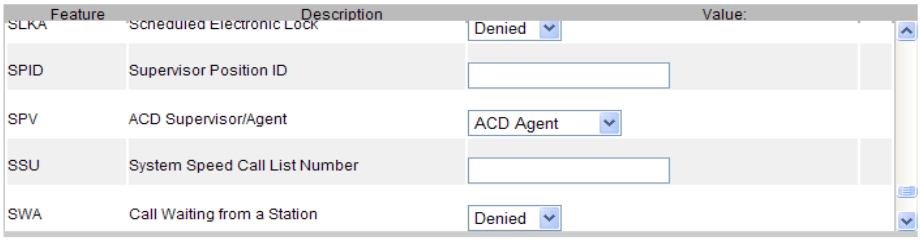
Step 2 – Click on the TN value displayed (e.g., **096 0 01 17**). The **Phone Details** form will open. Note that in this example the telephone type is an 1150 and that it is defined in Zone 3. A call between this telephone and another telephone in Zone 3 will use a "best bandwidth" strategy (see **Section 5.4**) and therefore can use G.711MU. If this same telephone connects to the PSTN via the SIP trunk, the call would use a "best bandwidth" strategy, and the call would use G.729A.



5.7.1.2 Features

Scroll further down the **Phone Details** form and locate the **Features** section of the form. In this section various CS1000E telephone features are defined. The feature described below is found by scrolling through this section.

Step 3 – For the **SPV - ACD Supervisor/Agent** field select **ACD Agent**.



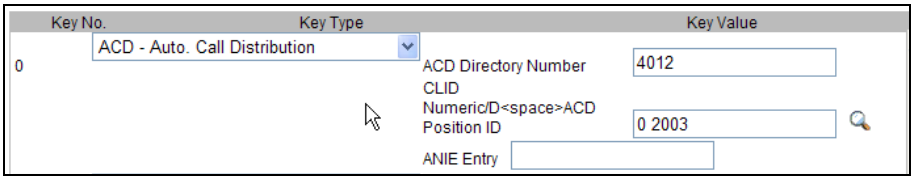
5.7.1.3 Keys

Scroll further down the **Phone Details** form and locate the **Keys** section of the form. Phone key positions (buttons) are defined in this section.

5.7.1.3.1 Key 0

Step 4 – For Key **0** select **ACD – Auto Call Distribution**

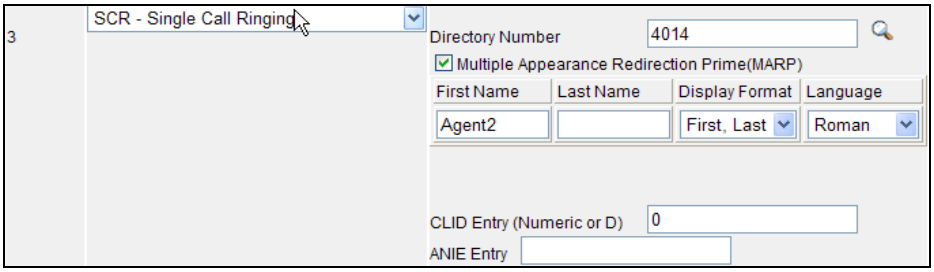
- For **ACD Directory Number** enter **4012**
- For **Numeric/D<space>ACD Position ID** enter **0 2003**



5.7.1.3.2 Key 3 - Single Call Appearance

Step 5 – For Key 3 select **SCR - Single Call Ringing**

- For **Directory Number** select **4014**
- Check **Multiple Appearance Redirection Prime(MARP)**
- Enter a name (e.g., Agent2)
- In the **CLID Entry** field, enter the associated CLID defined in **Section 5.8** (e.g., **0**).



5.7.1.3.3 LD 20 Overlay Command for Agent Configuration Display

The following CS1000E overlay command may be used to display/verify the Agent configuration.

```
OVL000
>ld 20
REQ: prt
TYPE: 1150
TN
CUST 0
DATE
PAGE
DES
MODEL_NAME
EMULATED
KEM_RANGE
DES  AGENT2
TN   096 0 01 17  VIRTUAL
TYPE 1150
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 00003
CUR_ZONE 00003
MRT
ERL  0
ECL  0
FDN
TGAR 1
LDN  NO
NCOS 0
SGRP 0
RNPG 0
SCI  0
SSU
XLST
SCPW
CLS  CTD FBD WTA LPR MTD FND HTD TDD HFA CRPD
     MWD LMPN RMMD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LND CNDA
     CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
     ICDD CDMD LLCN MCTD CLBD AUTU
     GPUD DPUD DNDA CFXD ARHD CNTD CLTD ASCD
     CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD AHD
     DDGA NAMA
     DRDD EXR0
     USMD USRD ULAD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
     VOLA VOUD CDMR PRED RECD MCDD T87D SBMD
     KEM3 MSNV FRA  PKCH MUTA MWTD DVLD CROD ELCD
CPND_LANG ENG
HUNT
```

```
PLEV 02
PUID
UPWD
DANI NO
SPID NONE
AST  00 03
IAPG 1
AACS YES
ACQ  AS: TN,AST-DN,AST-POSID
ASID 17
SFNB  1  2  3  4  5  6  7  8  9  10  11  12  13  15  16  17  18  19  22  24
SFRB  1  2  15
USFB  1  2  3  4  5  6  7  9  10  11  12  13  14  15
CALB  0  1  3  4  5  6  8  9  10  11  12
FCTB  1
ITNA NO
DGRP
PRI  01
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY  00 ACD 4012 0  2003
     AGN
     01 NRD
     02 MSB
     03 SCR 4014 0     MARP
     04
     05
     06
     07
     08
     09
     10 TRN
     11 AO6
     12 CFW 16
     13 RGA
     14 PRK
     15 RNP
     16
     17 PRS
     18 CHG
     19 CPN
     20
     21
```

5.7.2. CS1000E Auto Call Distribution (ACD)

The ACD information may be displayed by using the **ld 23** overlay command.

```
>ld 23
ACD000
MEM AVAIL: (U/P): 98772443    USED U P: 4778038 101868    TOT: 103652349
DISK SPACE NEEDED: 72 KBYTES
ACD DNS                AVAIL: 23992    USED:     8    TOT: 24000
REQ  prt
TYPE acd
CUST 0
ACDN 4012
MWC  NO
DSAC NO
MAXP 5
SDNB NO
BSCW NO
ISAP NO
AACQ NO
RGAI NO
ACAA NO
FRRT
SRRT
NRRT
FROA NO
CALP POS
ICDD NO
NCFW
FNCF NO
FORC NO
RTQT 0
SPCP NO
OBTN NO
RAO  NO
CWTH 1
NCWL NO
BYTH 0
OVTH 2047
TOFT NONE
HPQ  NO
OCN  NO
OVDN
IFDN
OVBU LNK LNK LNK LNK
EMRT
MURT
RTPC NO
HOML YES
RDNA NO
LABEL_KEY0 NO
ACNT
NRAC NO
```

```
DAL  NO
RPRT YES
RAGT 4
DURT 30
RSND 4
FCTH 20
CRQS 100
SIPQ NO
IVR  NO
OBSC NO
OBPT 5
CWNT NONE
```

5.7.3. CS1000E Control DN (CDN)

The CDN information may also be displayed by using the **ld 23** overlay command.

```
>ld 23
ACD000
MEM AVAIL: (U/P): 98772394    USED U P: 4778038 101917    TOT: 103652349
DISK SPACE NEEDED: 71 KBYTES
ACD DNS                AVAIL: 23992    USED:     8    TOT: 24000
REQ  prt
TYPE cdn
CUST 0
CDN  4013
FRRT
SRRT
FROA NO
UUI  NO
MURT
CDSQ NO
DFDN 4012
NAME NO
CMB  NO
CEIL 2047
CLRO NO
OVFL NO
TDNS NO
RPRT YES
AACQ YES
ASID 17
SFNB 33  35  36  37  38  39
USFB 1  3  4  5  6  7  9  10  11  12  13  14  15
CALB 0  1  2  3  4  5  6  8  9  10  11  12
CNTL YES
VSID
HSID
CWTH 1
BYTH 0
OVTH 2047
ACNT
```

5.7.4. Analog Fax Line

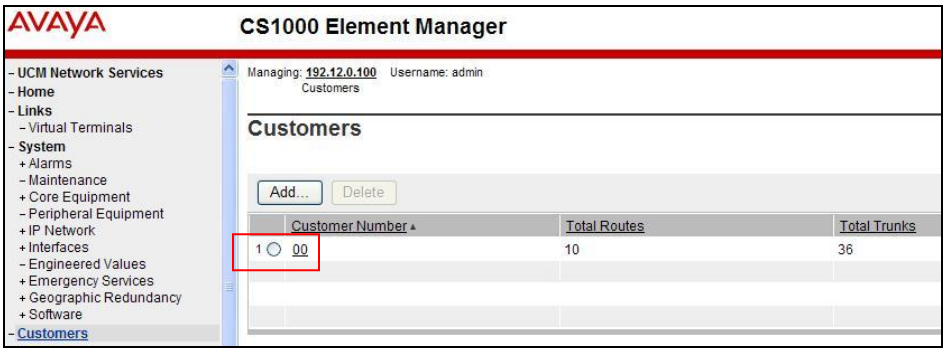
The following screen shows basic information for an analog port in the configuration that may be used with a fax machine. The port is configured as Directory Number 2779. No special Features or Keys were defined.

5.8. Customer Information

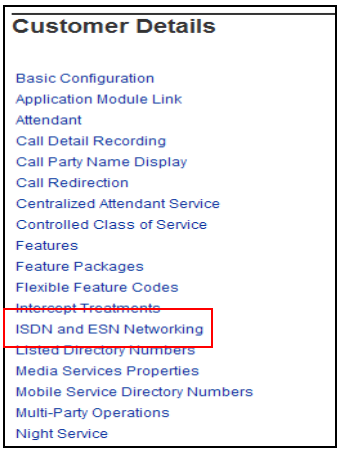
In the reference configuration, specific calling number information is required based on the destination of the call.

5.8.1. Caller ID Provisioning

Step 1 - Select **Customers** from the left navigation menu, click on the appropriate **Customer Number** (e.g., **00)**



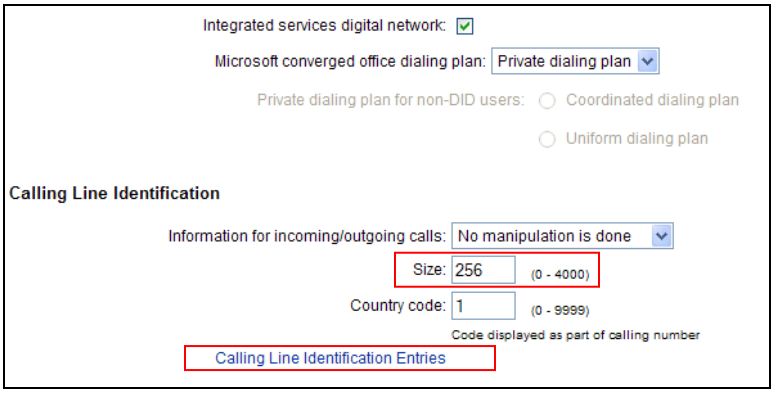
Step 2 – The Customer Details screen will open. Select **ISDN and ESN Networking**.



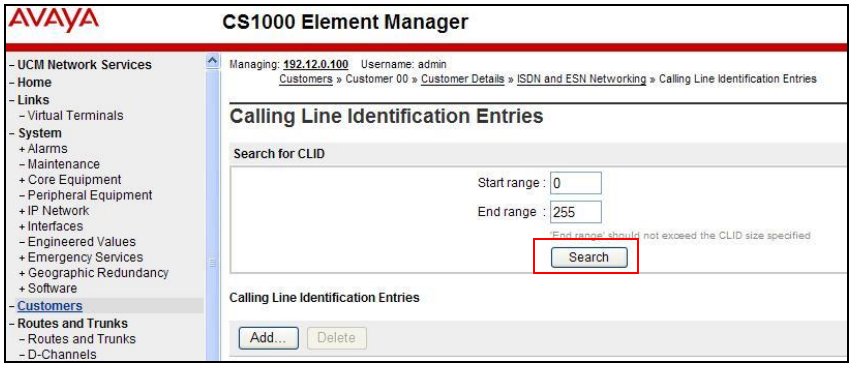
The ISDN and ESN Networking **General Properties** screen will open (not shown).

Step 3 - Scroll down from **General Properties** to the **Calling Line Identification** section of the page and note the value in the **Size** parameter (e.g., **256**).

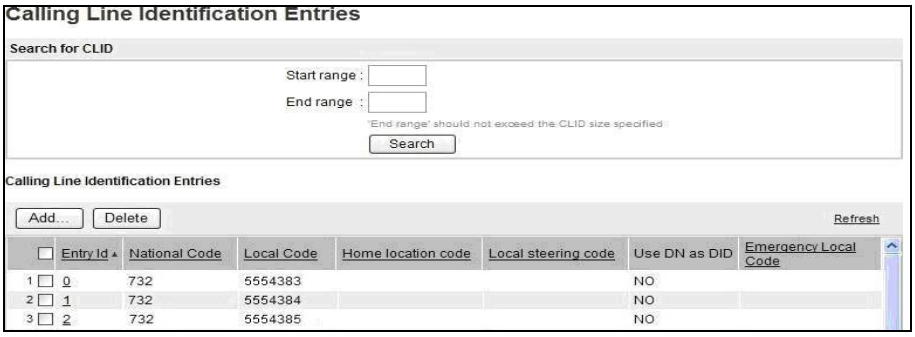
Step 4 - Click the **Calling Line Identification Entries** link.



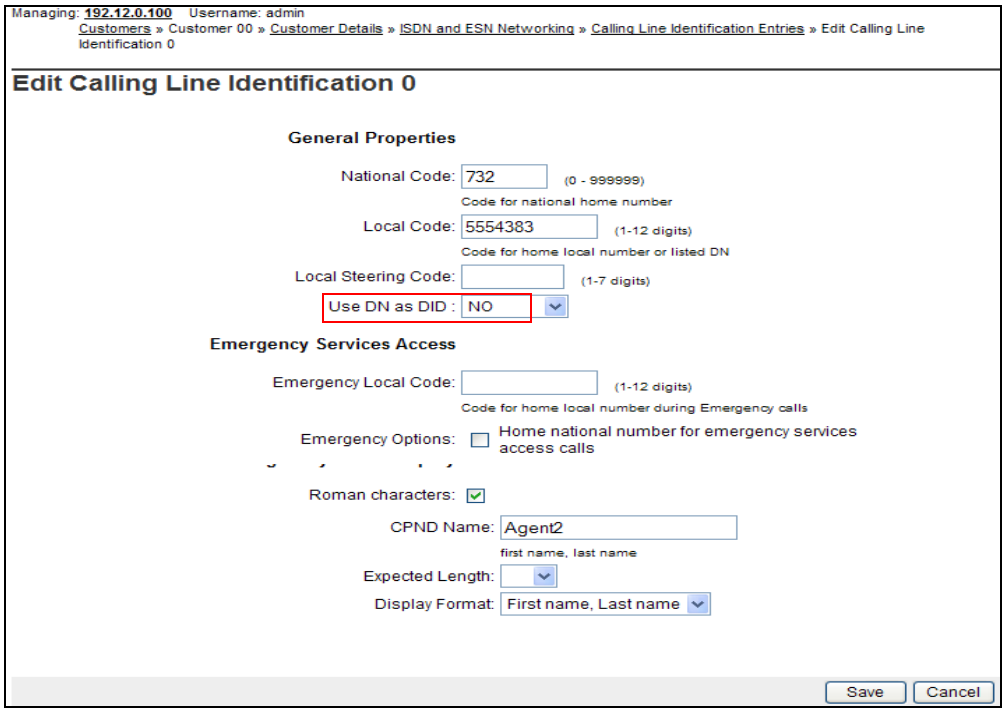
Step 5 – In the **Search for CLID** section, enter **0** in the **Start range** field and in the **End range** field enter one less than the **Size** value from **Step 3** above (e.g., enter **255**). Click on **Search**.



This will display all defined Call Ids. For example **CLID 0** will use **732-555-4383**.



Click on any Entry ID to view or change further details (e.g., **Entry ID 0**). Note that the **Use DN as DID** is set to **NO**. This means that the local extension will not be used for the calling number. Call IDs are then associated with specific telephone directory numbers (DNs) assigned to stations (see **Section 5.7.1.3.2**).



5.9. Changing RFC2833 DTMF Telephone Event Type

The CS1000E uses RFC2833 DTMF Telephone Event type 101. The AT&T IP Toll Free service recommends the value 100. While having asymmetric telephone event types is permitted, this may cause issues in some call scenarios. Therefore the CS1000E value may be changed to 100 as follows:

Step 1 – From an CS1000E console connection, press the ctrl key and enter "**pdt**". The system will return:

```
PDT login on /tyCo/0
Username:
```

Step 2 – Enter the appropriate username. The system will respond with:

```
Password:
```

Step 3 – Enter the appropriate password. The system will respond as follows:

```
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>
```

Step 4 – At the pdt> prompt enter "**setRFC2833PT 100**"

```
pdt> setRFC2833PT 100
```

The system will respond with the pdt> prompt.

```
pdt>
```

The CS1000E will now use RFC2833 DTMF telephone event type 100.

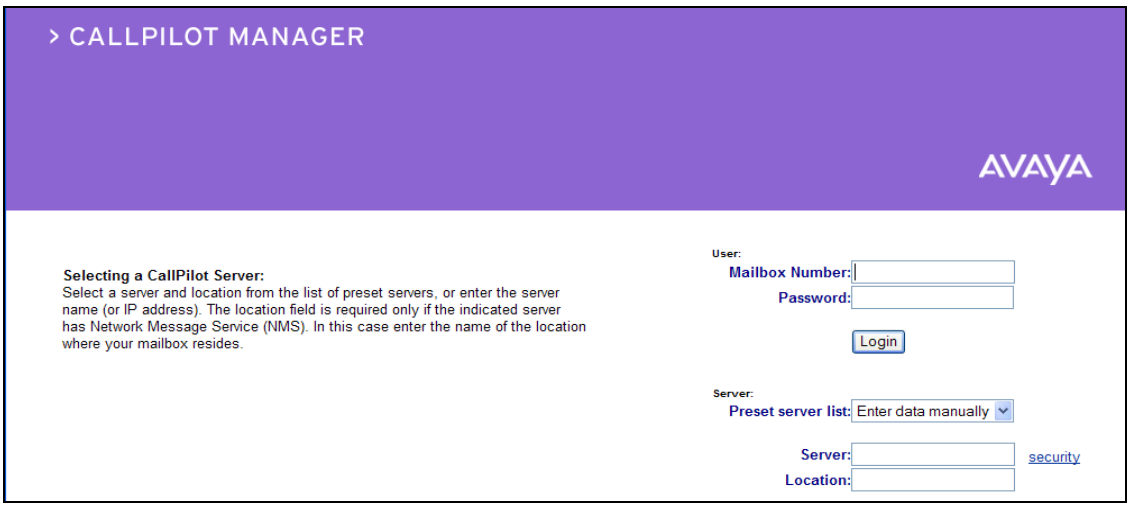
Note – If the CS1000E is rebooted, this command will be cleared and the system will use telephone event 101 again. This command must be re-entered.

5.10. Inbound Calls to Call Pilot®

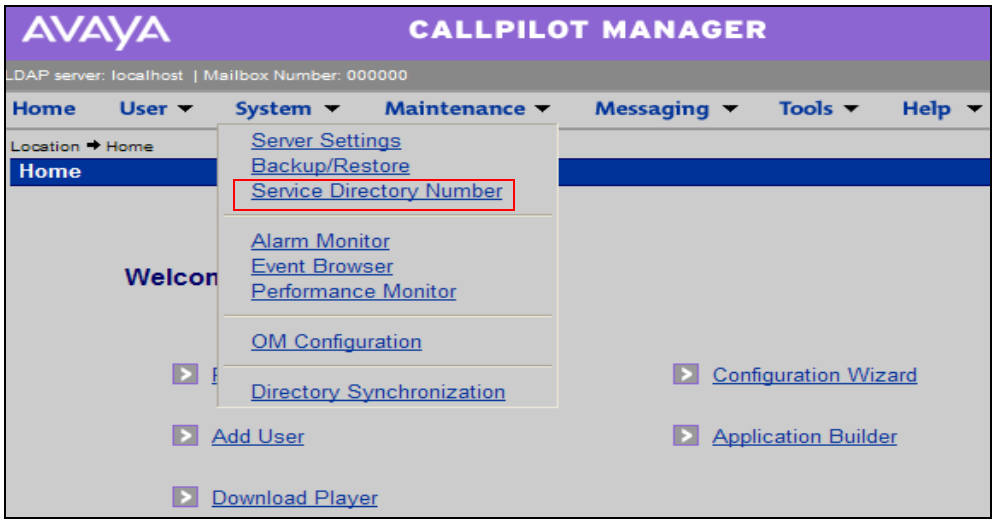
PSTN callers may wish to access Call Pilot® to retrieve messages. In addition to defining an entry in the CS1000E IDT table for routing calls to the main Call Pilot® access number (e.g., 2090, see **Section 5.6**), the customers Billing Number (that the AT&T IP Toll Free service inserts in SIP INVITE Request-URI and TO headers, see **Section 2.2.1, Item 5**), must be defined to Call Pilot® as well. This is required because Call Pilot® uses the contents of the TO header for admission control.

Note – The provisioning of Call Pilot® is beyond the scope of this document. Refer to [**5**] for more information.

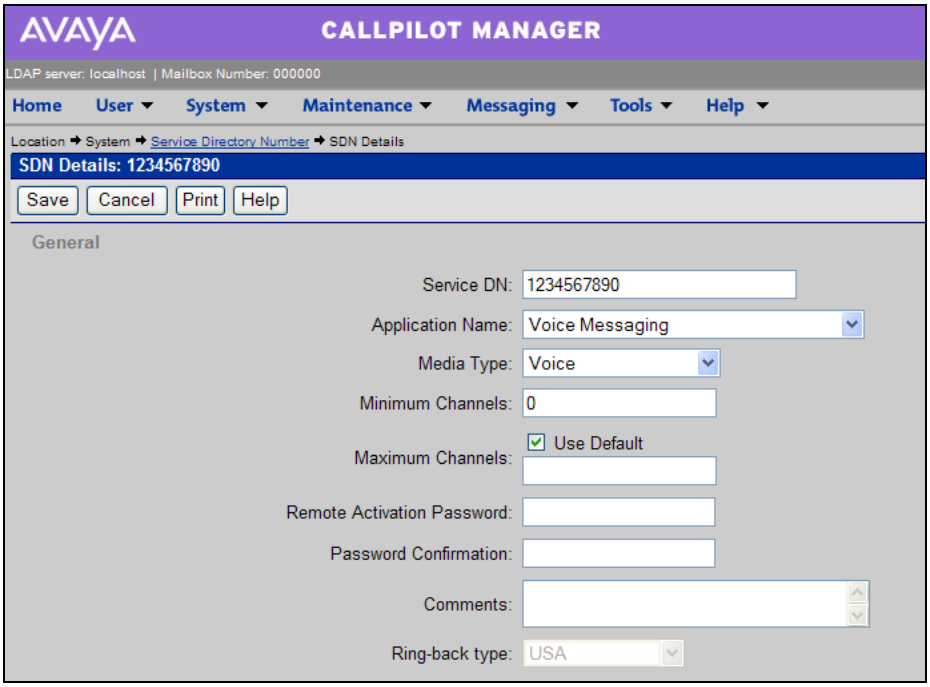
Step 1 – Log into the Call Pilot® manager GUI using the appropriate credentials.



Step 2 – Navigate to **System → Service Directory Number**

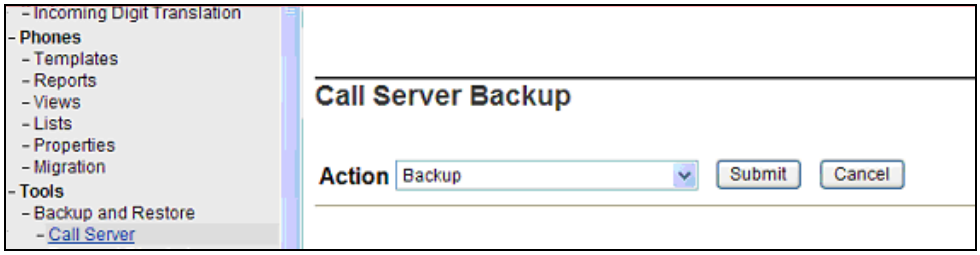


Step 3 – Click on **New** (not shown). Populate the form as shown below, where **1234567890** is the AT&T IP Toll Free customer Billing Number. Click on **Save**.



5.11. Configuration Backup

Expand **Tools → Backup and Restore** on the left navigation panel and select **Call Server**. Select **Backup** and click **Submit** to save configuration changes as shown below.



The backup process may take several minutes to complete. Scroll to the bottom of the page to verify the backup process completed successfully as shown below.

6. Configure Avaya Aura® Session Manager

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

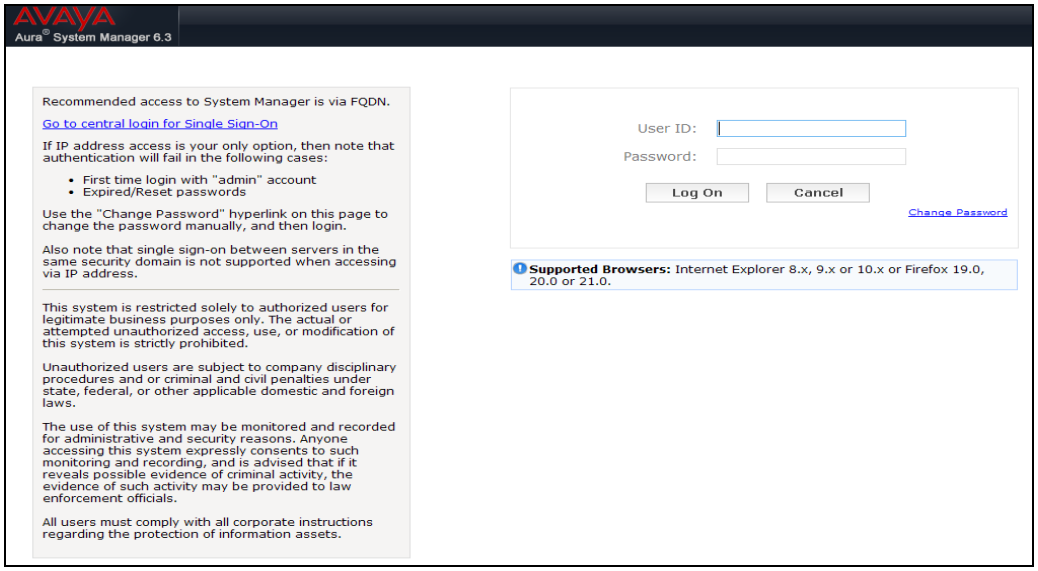
Note – The following sections assume that Session Manager and System Manager have been installed and that network connectivity exists between the two. For more information, consult the references in **Section 11**.

The following administration activities will be described:

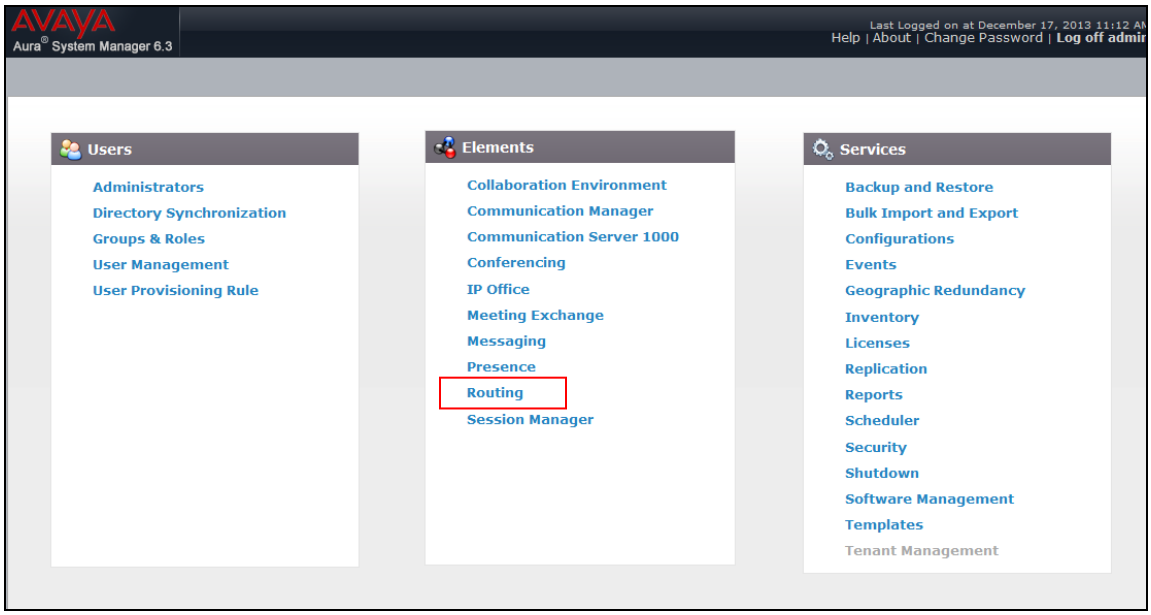
- Define SIP Domain
- Define Locations for CS1000E and for the Avaya SBCE
- Configure the Adaptation Modules that will be associated with the SIP Entities for CS1000E and the Avaya SBCE
- Define SIP Entities corresponding to CS1000E and Avaya SBCE
- Define Entity Links describing the SIP trunk between CS1000E and Session Manager, and the SIP trunk between Session Manager and Avaya SBCE.
- Define Routing Policies associated with CS1000E and Avaya SBCE.
- Define Dial Patterns, which govern which Routing Policy will be selected for call routing.

Step 1 - Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL "**http://<ip-address>/SMGR**", where **<ip-address>** is the IP address of System Manager. Log in with the appropriate credentials.

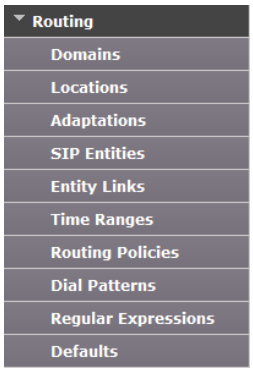
Step 2 - From the **Log On** screen enter appropriate **User ID** and **Password** and press the **Log On** button.



Step 3 - Once logged in, a Release 6.3.5 **Home** screen like the following is displayed. From the **Home** screen below, under the **Elements** heading in the center, select **Routing**.



The screen shown below shows the various sub-headings of the left navigation menu that will be referenced in this section.



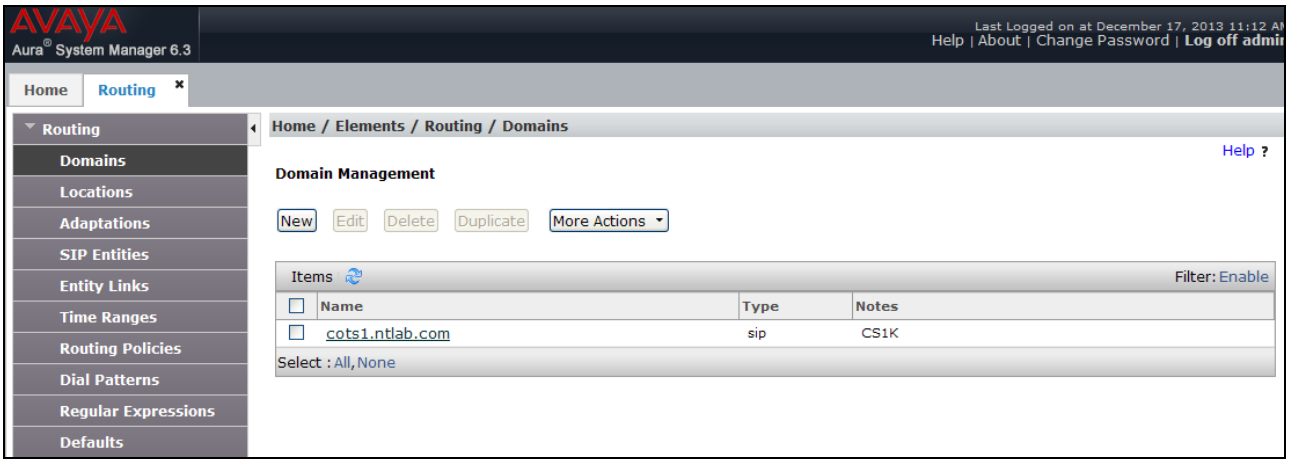
6.1. SIP Domain

Step 1 - Select **Domains** from the left navigation menu. In the reference configuration domain "cots1.ntlab.com" was defined.

Step 2 - Click **New** (not shown). Enter the following values shown below and use default values for remaining fields.

- **Name** = **cots1.ntlab.com** (see **Section 5.5.1**).
- **Type** = SIP
- Add **Notes** if desired.

Step 3 - Click **Commit** to save (not shown). Multiple SIP Domains may be defined if required.



6.2. Locations

Locations are used to identify logical and/or physical locations where SIP Entities reside. Location identifiers can be defined in a broad scope (e.g., 172.16.6.x for all devices on a particular subnet), or individual devices (e.g., 172.16.6.10 for a specific device's IP address). In the reference configuration the CS1000E is located in subnet 172.16.6.x). The rest of the CPE equipment, (including Session Manager, System Manager and the Avaya SBCE), were located in subnet 192.168.67.x. Therefore a Location was created for each subnet.

6.2.1. Location for CS1000E Subnet

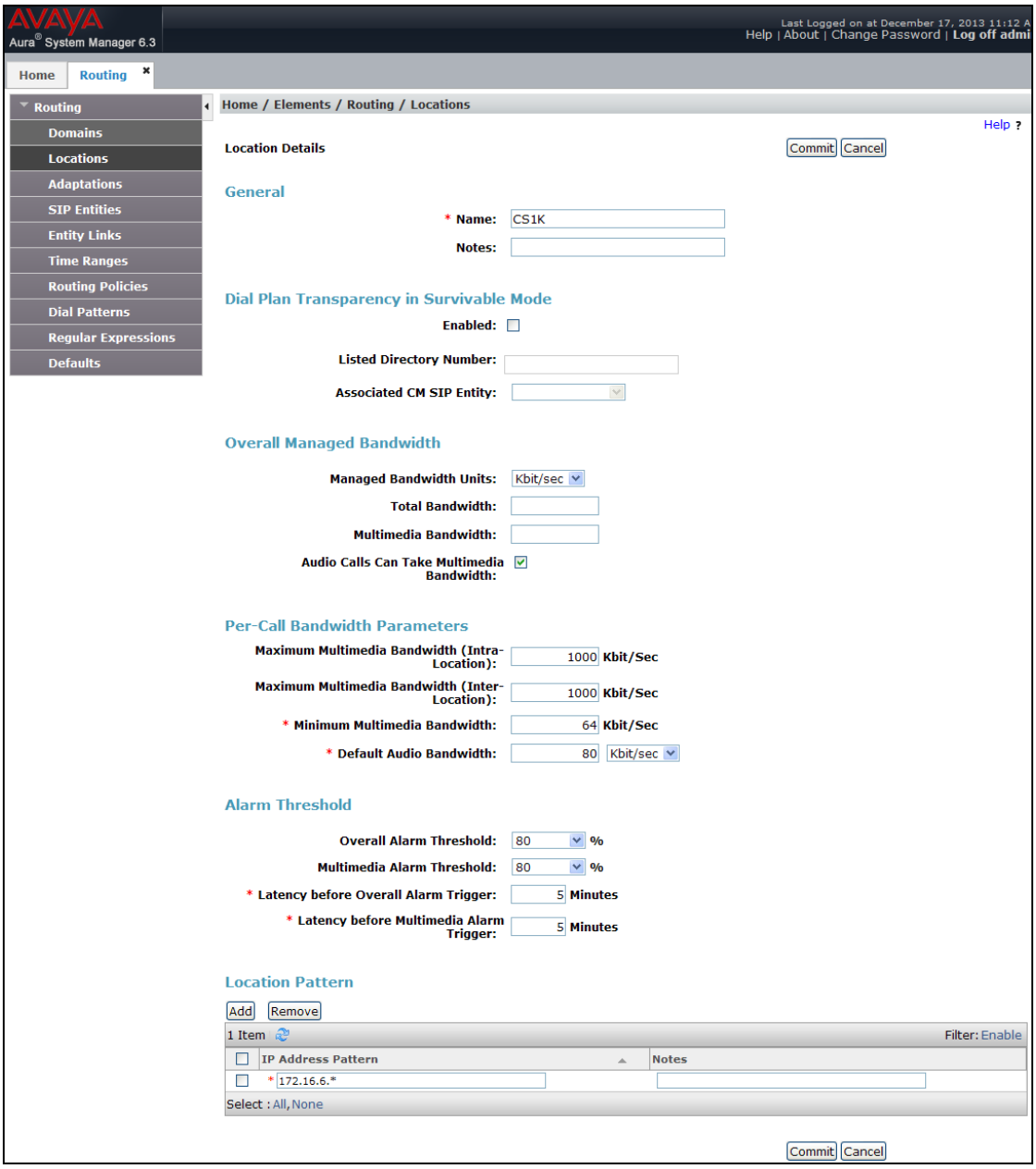
Step 1 - Select **Locations** from the left navigational menu. Click **New** (not shown)**.** In the **General** section**,** enter the following values and use default values for remaining fields.

- **Name:** Enter a descriptive name for the location (e.g., **CS1K**).
- **Notes:** Add a brief description. [Optional]

Step 2 - In the **Location Pattern** section, click **Add** and enter the following values.

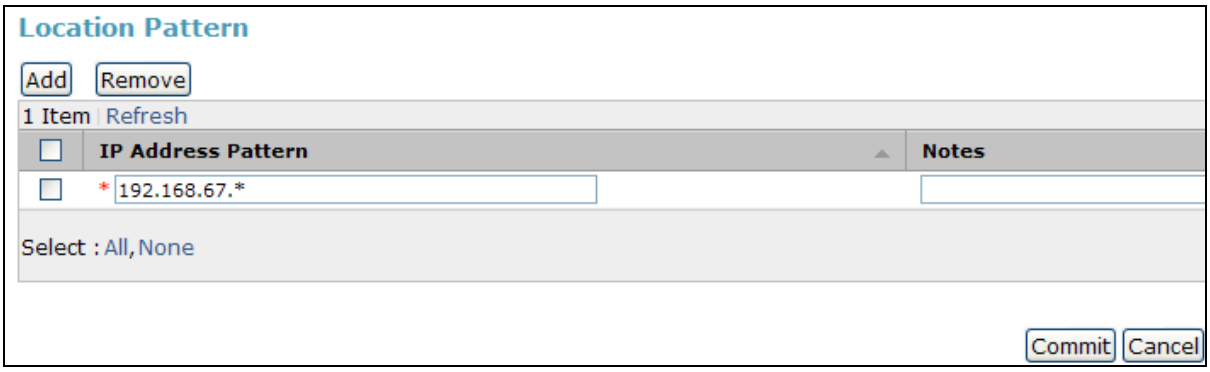
- **IP Address Pattern** Enter the IP Address or IP Address pattern used to identify the CS1000E location (e.g., **172.16.6.***).
- **Notes** Add a brief description. [Optional]

Step 3 – Leave all other default values, and click **Commit** to save.

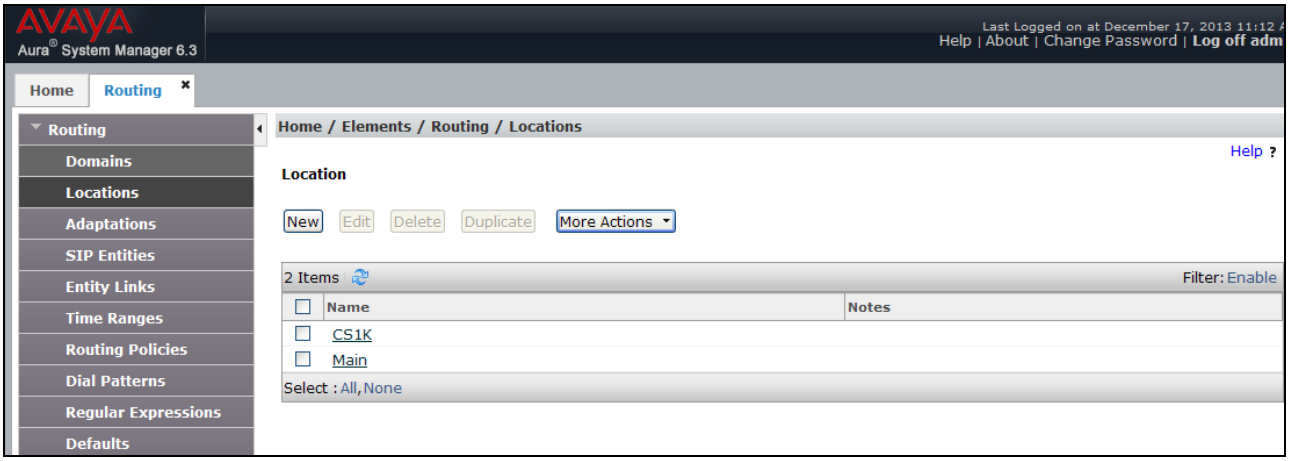


6.2.2. Location for Customer Premises Equipment Subnet.

Repeat **Steps 1-3** in **Section 6.2.1** to create a location called **Main** for the rest of the CPE, including Session Manager, System Manager, and the Avaya SBCE, using address **192.168.67.***.



The completed Locations form is shown below.



6.3. Configure Adaptations

Adaptations are pre-configured modules, designed for the CS1000E, that Session Manager can use to replace, modify, or remove SIP headers. In the reference configuration the following adaptations are used.

- **CS1000Adapter** – This adaptation is used to provide translation of certain CS1000E generated SIP headers into formats used by other Avaya products and endpoints.
- **DigitConversionAdapter** – This adaptation is used in conjunction with the CS1000Adapter to modify digit strings in the SIP Request-URI. Note that this adaptation's functionality is included in all other adaptations.

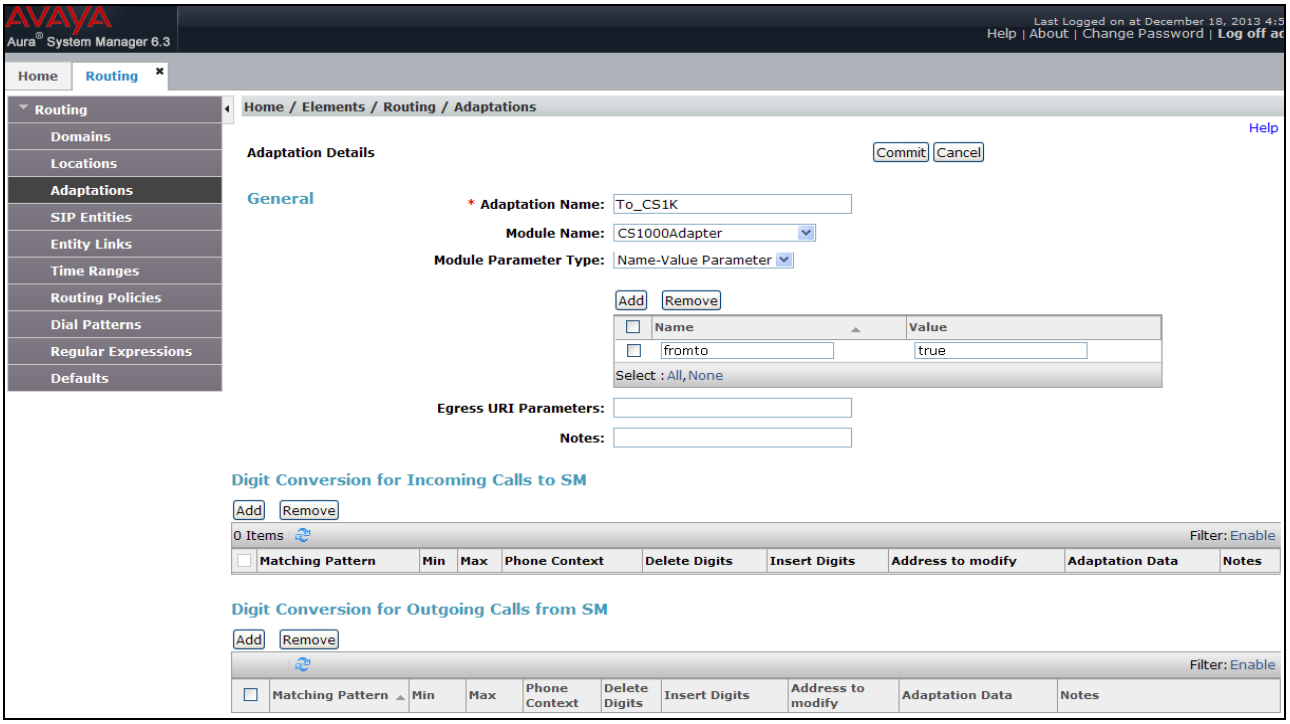
In addition, Module Parameters **fromto=true** (used to modify the From and To headers for inbound calls to Call Pilot), and **MIME=no** (to remove CS1000E Mime headers not supported by AT&T), are also specified.

6.3.1. Adaptation for Traffic to CS1000E

Step 1 - Select **Adaptations** from the left navigational menu. Click **New (**not shown**)**. In the **General** section, enter the following values and use default values for remaining fields.

- **Adaptation Name:** Enter an identifier for the Adaptation Module (e.g., "**To_CS1K**").
- **Module Name:** Select "**CS1000Adapter**" from drop-down menu (or enter "**CS1000Adapter**" if not previously defined in the menu).
- **Module Parameter:** Select **Name-Value Parameter** from the drop-down menu.
  - Click on **Add**, then enter:
    - **Name** = **fromto**
    - **Value** = **true**

Step 2 - Click on **Commit.**



Note – No entries are required in the **Digit Conversion for Incoming Calls to SM** or the **Digit Conversion for Outgoing Calls from SM** sections.

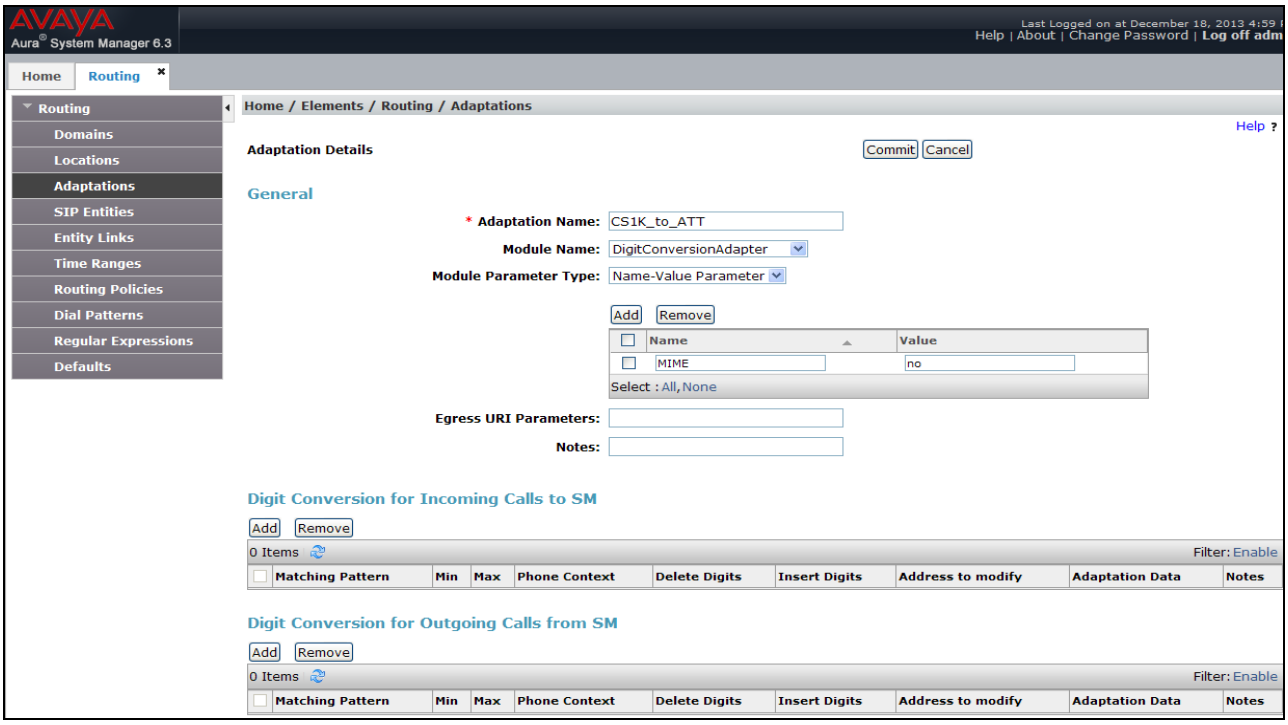
6.3.2. Adaptation for Traffic from CS1000E to AT&T

The message body of Re-INVITE messages sent from the CS1000E may contain a MIME Multipart message body containing the SDP information expected by AT&T, but also containing "x-nt-mcdn-frag-hex" and "x-nt-epid-frag-hex" application parts that are not processed by AT&T. Since AT&T has no use for this information, the Module Parameter **MIME=no** was used in the reference configuration to remove these headers. Note that the Avaya SBCE is used to remove additional Avaya SIP headers (see **Section 8.4.3**).

Step 1 - Select **Adaptations** from the left navigational menu. Click **New (**not shown**).** In the **General** section, enter the following values and use default values for remaining fields**.**

- **Adaptation Name:** Enter an identifier for the Adaptation Module (e.g., "**CS1K_to_ATT**").
- **Module Name:** Select "**DigitConversionAdapter**" from drop-down menu (or add an adapter with name "**DigitConversionAdapter**" if not previously defined)
- **Module Parameter Type:** Select **Name-Value Parameter** from the drop-down menu.
  - Click on **Add**, then enter:
    - **Name** = **MIME**
    - **Value** = **no**

Step 2 - Click **Commit.**



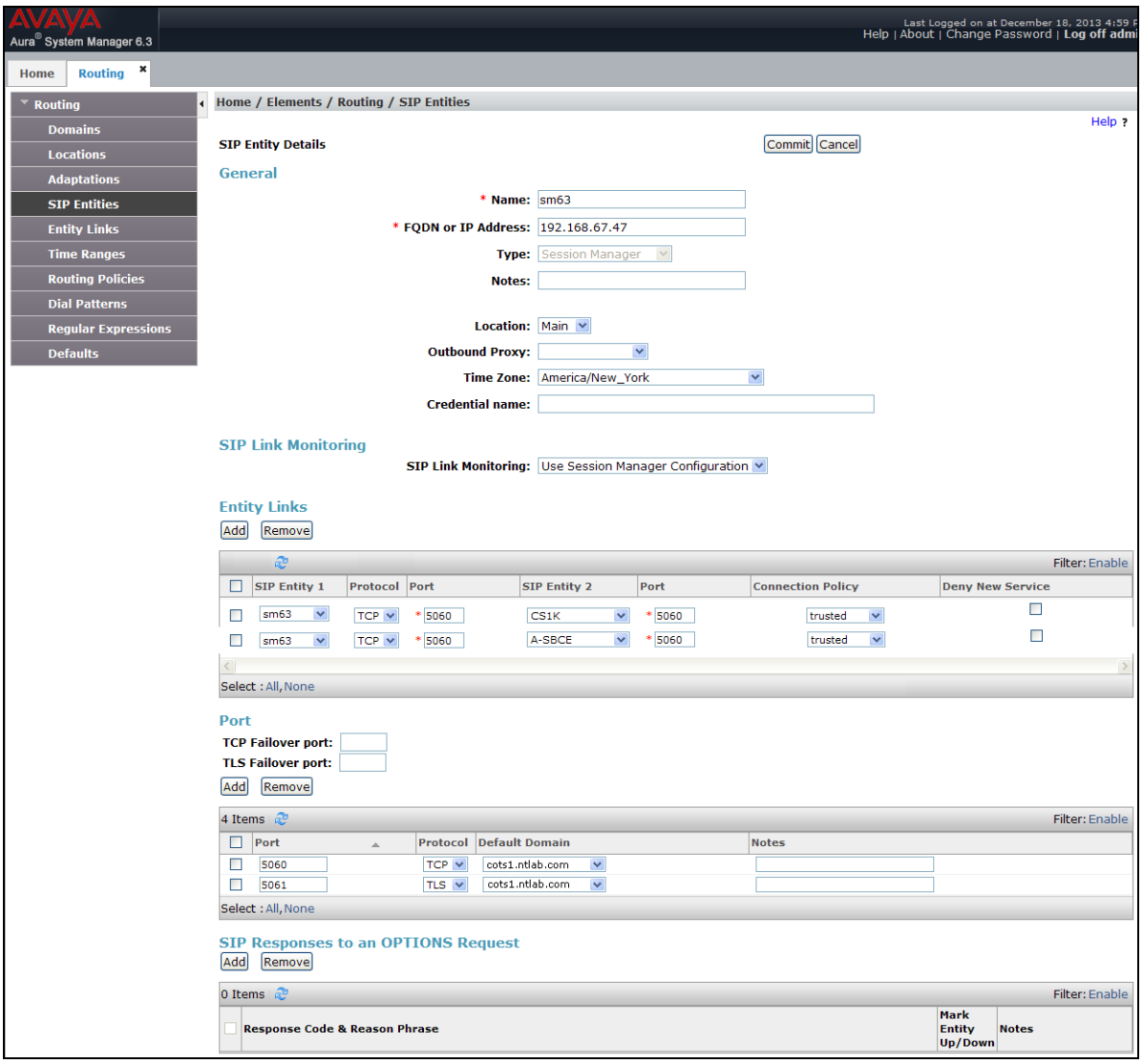
Note – No entries are required in the **Digit Conversion for Incoming Calls to SM** or the **Digit Conversion for Outgoing Calls from SM** sections.

6.4. SIP Entities

SIP Entities must be added for CS1000E and Avaya SBCE. Note that once Entity Links are provisioned for each Entity (see **Section 6.5**), the Entity Link information will also be displayed on the Entity forms.

In addition, a SIP Entity is also created for Session Manager itself, during the System Manager installation process. While installation procedures are beyond the scope of this document, the Session Manager Entity form is shown below for completeness. Consult [**6-8**] for further information on Session Manager Installation.

6.4.1. SIP Entity for Session Manager



6.4.2. SIP Entity for CS1000E

Step 1 - Select **SIP Entities** from the left navigation menu.

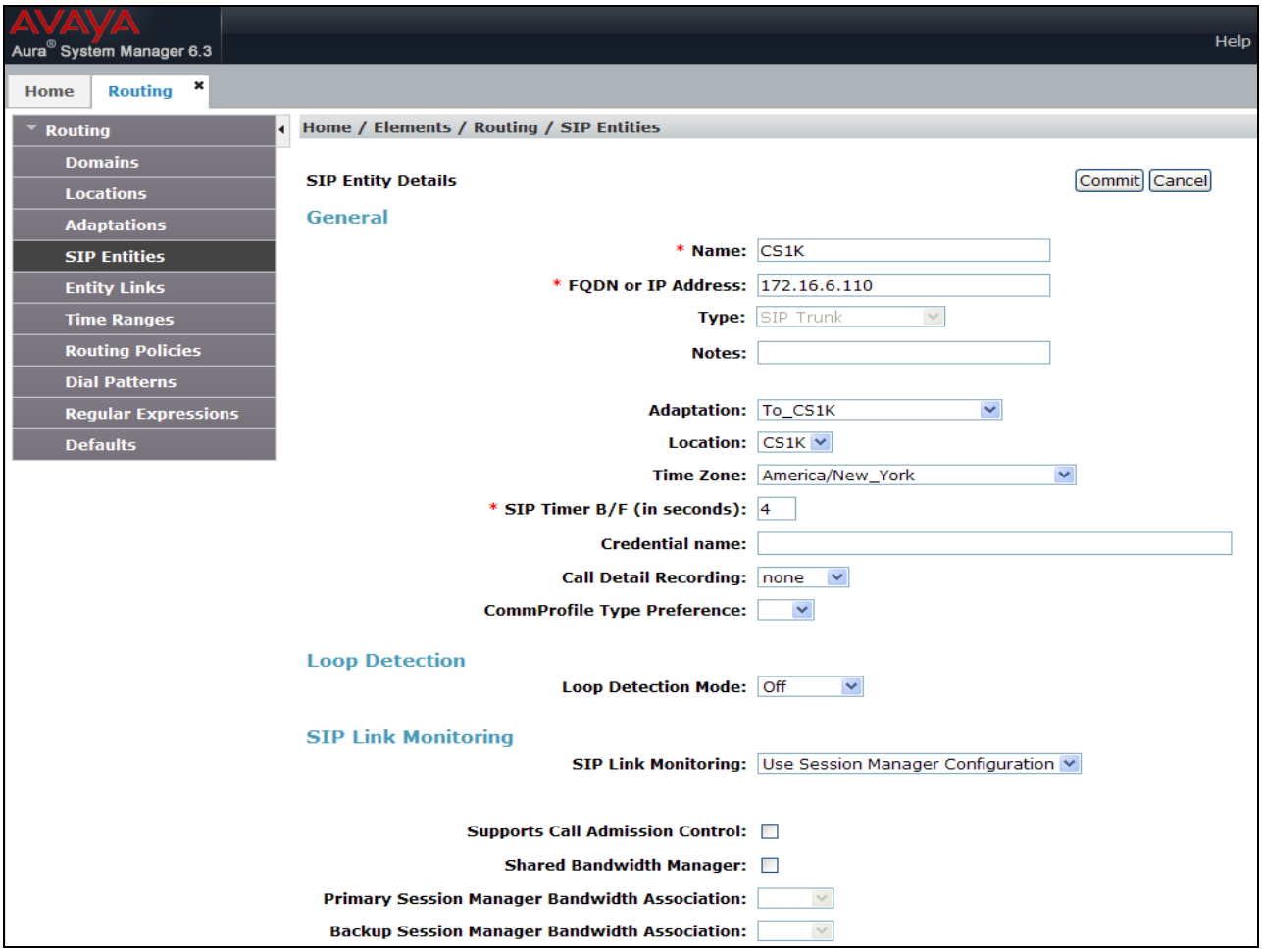
Step 2 - Click **New (**not shown**).** In the **General** section, enter the following values and use default values for remaining fields**.**

- **Name:** Enter an identifier for the SIP Entity (e.g., "**CS1K**").
- **FQDN or IP Address:** Enter the TLAN IP address of the CS1000E SIP GW.
- **Type:** Select "**SIP Trunk**"
- **Notes:** Enter a brief description. [Optional]
- **Adaptation:** Select the **To_CS1K** Adaptation Module defined in **Section 6.3.1**.
- **Location:** Select the **CS1K** Location defined in **Section 6.2.1**.

Step 3 - In the **SIP Link Monitoring** section:

- **SIP Link Monitoring:** Select "**Use Session Manager Configuration**" (or choose an alternate Link Monitoring approach for this Entity, if desired).

Step 4 - Click **Commit**.



6.4.3. SIP Entity for Avaya SBCE

Step 1 - Select **SIP Entities** from the left navigation menu.

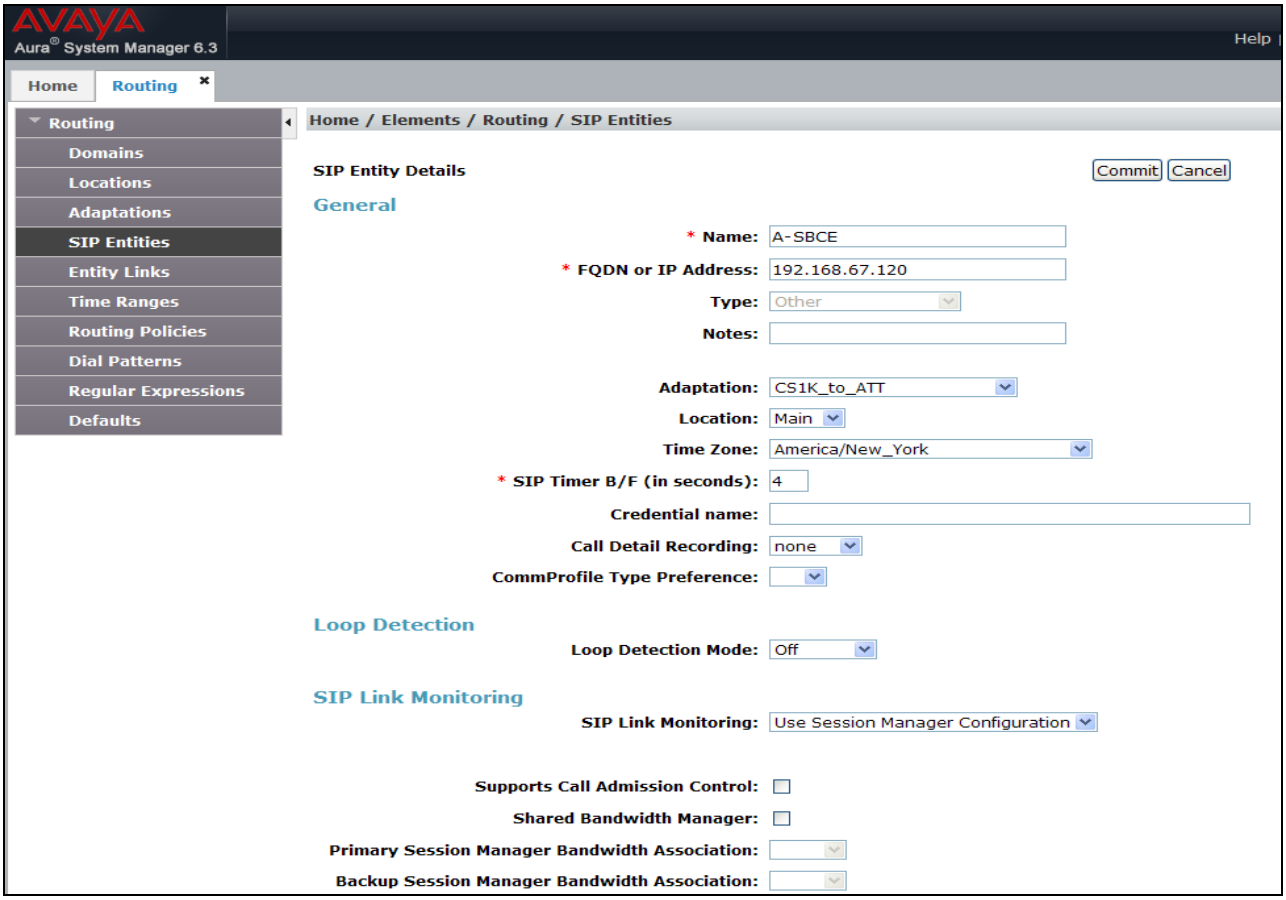
Step 2 - Click **New (**not shown**).** In the **General** section, enter the following values and use default values for remaining fields.

- **Name:** Enter an identifier for the SIP Entity (e.g., "**A-SBCE**").
- **FQDN or IP Address:** Enter the IP Address of the Avaya SBCE private side (A1) interface.
- **Type:** Select "**Other**"
- **Notes:** Enter a brief description. [Optional]
- **Adaptation:** Select the **CS1K_to_ATT** Adaptation Module defined in **Section 6.3.2**.
- **Location:** Select the Location **Main** defined in **Section 6.2.2**.

Step 3 - In the **SIP Link Monitoring** section:

- **SIP Link Monitoring:** Select "**Use Session Manager Configuration**" (or choose an alternate Link Monitoring approach for this Entity, if desired).

Step 4 - Click **Commit**.



6.5. Entity Links

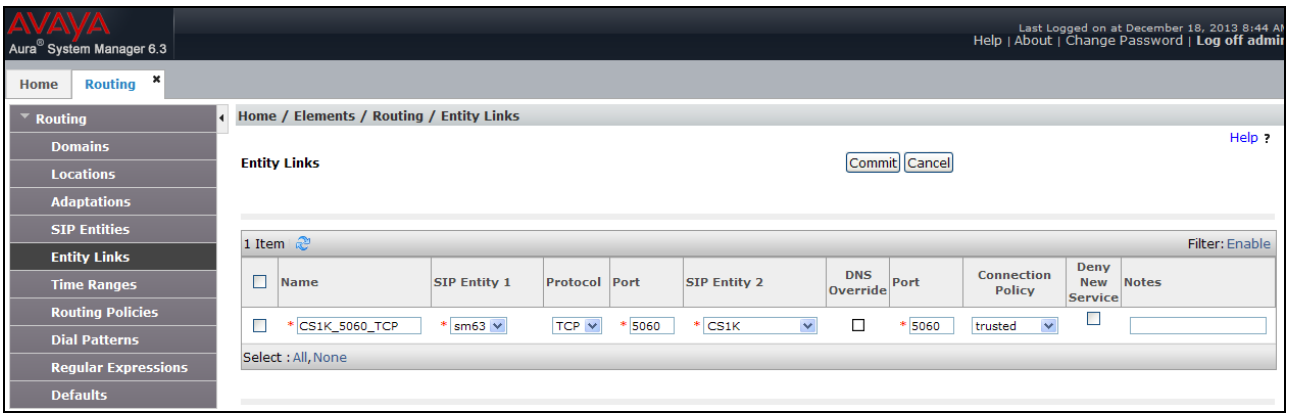
The SIP trunk between Session Manager and CS1000E is defined by an Entity Link, as is the SIP trunk between Session Manager and Avaya SBCE.

6.5.1. Entity Link to CS1000E

Step 1 - Select **Entity Links** from the left navigation menu.

Step 2 - Click **New** (not shown). Enter the values shown below.

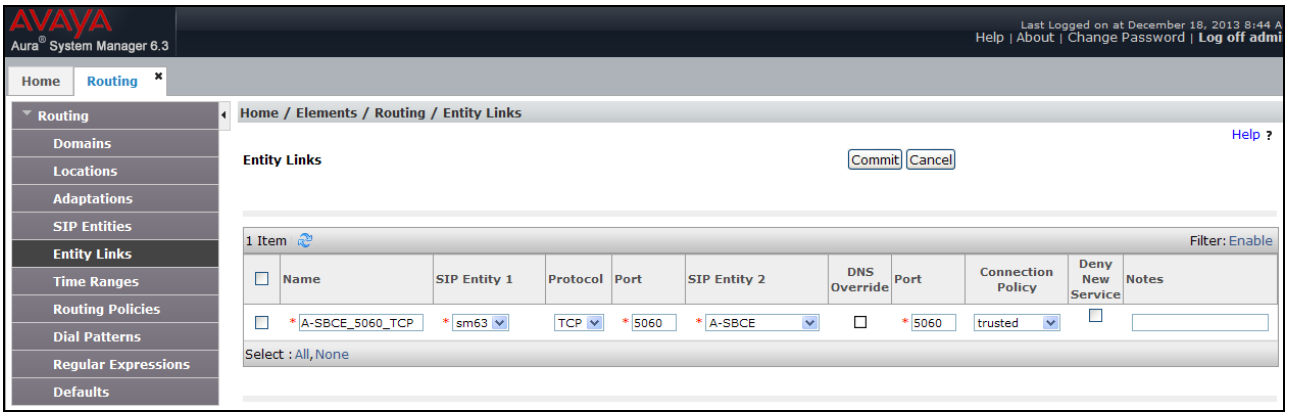
Step 3 - Click **Commit**.



6.5.2. Entity Link to Avaya SBCE

Step 1 - Select **Entity Links** from the left navigation menu. Click **New** (not shown). Enter the values shown below.

Step 2 - Click **Commit** to save the **Entity Link** definition.



6.6. Routing Policies

Routing Policies describe the conditions under which calls will be routed by Session Manager to CS1000E, or Avaya SBCE.

6.6.1. Routing Policy to CS1000E

Step 1 - To add a new Routing Policy, select **Routing Policies.** Click **New** (not shown). In the **General** section, enter the following values.

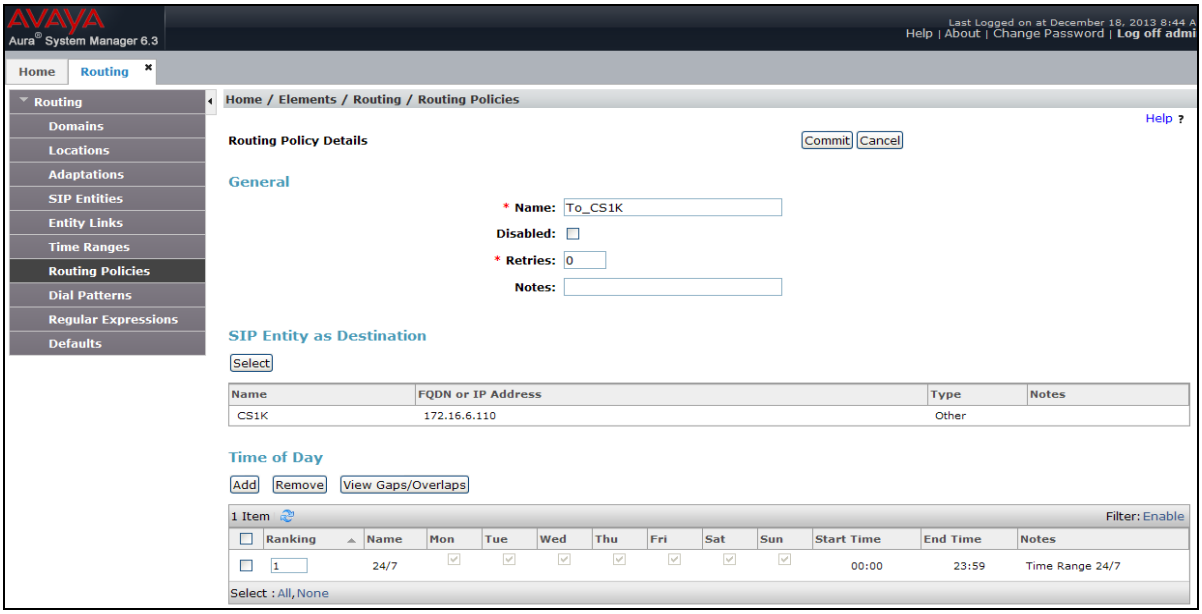
- **Name:** Enter an identifier to define the Routing Policy (e.g., "**To_CS1K**").
- **Disabled:** Leave unchecked.
- **Notes:** Enter a brief description. [Optional]

Step 2 - In the **SIP Entity as Destination** section, click **Select.** The **SIP Entity List** page opens (not shown).

- Select the **CS1K** SIP Entity associated with CS1000E (see **Section 6.4.2**) and click **Select**, and the display returns to the **Routing Policy Details** page.

Step 3 - In the **Time of Day** section, add an appropriate time of day entry. In the sample configuration, time of day was not a relevant routing criteria, so the "24/7" range was chosen. Use default values for remaining fields.

Step 4 - Click **Commit** to save the Routing Policy definition.



Note – The Dial Pattern portion of this form will be populated when the Dial Patterns in **Section 6.7** are defined.

6.6.2. Routing Policy to Avaya SBCE

Step 1 - To add a new Routing Policy, select **Routing Policies.** Click **New** (not shown). In the **General** section, enter the following values.

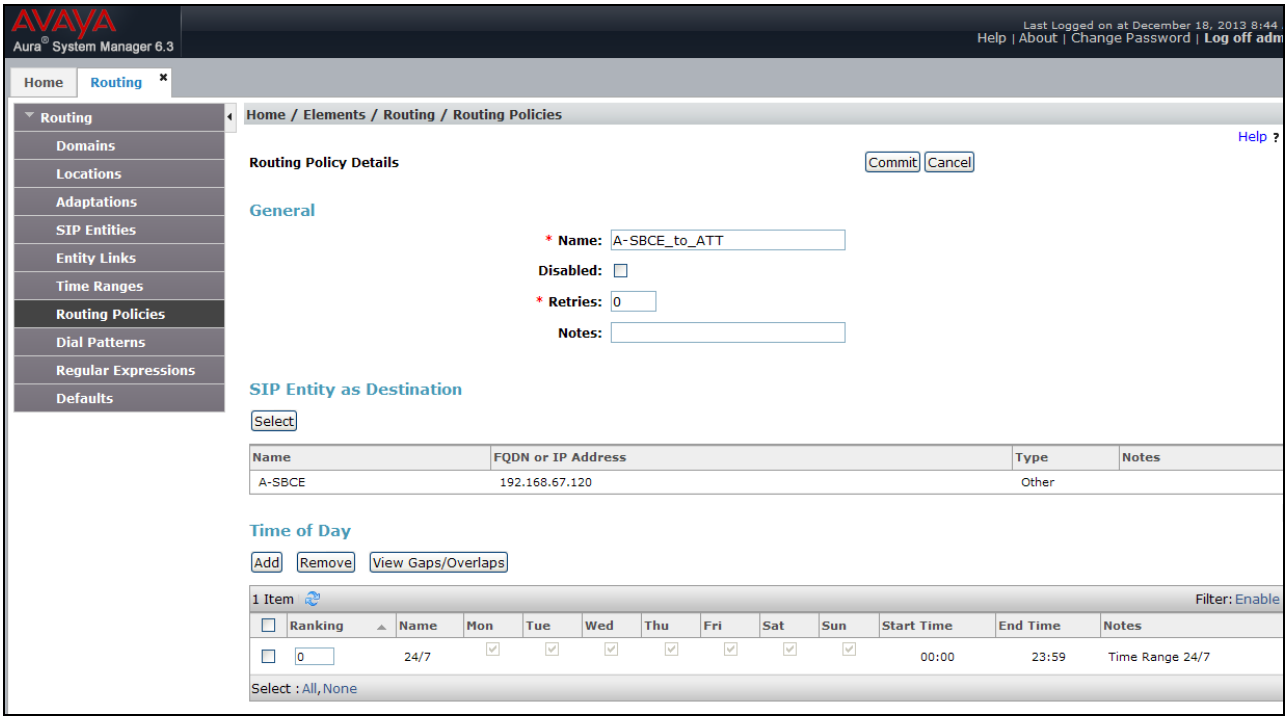
- **Name:** Enter an identifier to define the Routing Policy (e.g., "**A-SBCE_to_ATT**").
- **Disabled:** Leave unchecked.
- **Notes:** Enter a brief description. [Optional]

Step 2 - In the **SIP Entity as Destination** section, click **Select.** The **SIP Entity List** page opens (not shown).

- Select the **A-SBCE** SIP Entity associated with Avaya SBCE (see **Section 6.4.3**) and click **Select**, and the display returns to the **Routing Policy Details** page.

Step 3 - In the **Time of Day** section, add an appropriate time of day entry. In the sample configuration, time of day was not a relevant routing criteria, so the "24/7" range was chosen. Use default values for remaining fields.

Step 4 - Click **Commit** to save the Routing Policy definition.



Note – The Dial Pattern portion of this form will be populated when the Dial Patterns in **Section 6.7** are defined.

6.7. Dial Patterns

Dial patterns are used to route calls to the appropriate Routing Policies, and ultimately to the appropriate SIP Entities.

Note - The dialed AT&T DID numbers may not be the same as the AT&T DNIS numbers sent in the SIP Request-URI headers. The DNIS numbers used in the Request-URIs are the numbers to be defined here in the **Pattern** fields. In the examples below, an inbound 10 digit DNIS pattern of **732555xxxx** is used. These patterns are also matched, and converted to local extensions, by the CS1000E IDT table (see **Section 5.6**).

6.7.1. Inbound AT&T calls to CS1000E Extensions

Step 1 - To define a Dial Pattern, select **Dial Patterns** from the navigation menu. Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

- **Pattern:** Enter a matching dial pattern for calls to the CS1000E (e.g., **732555**)
- **Min:** Enter the minimum number of digits (e.g., 10).
- **Max:** Enter the maximum number of digits (e.g., 10).
- **SIP Domain:** Select a SIP Domain from drop-down menu or select **"All**" if Session Manager should route incoming calls from all SIP domains.
- **Notes:** Enter a brief description. [Optional]

Step 2 - In the **Originating Locations and Routing Policies** section, click **Add**.

Step 3 - The **Originating Locations and Routing Policy List** page opens (not shown).

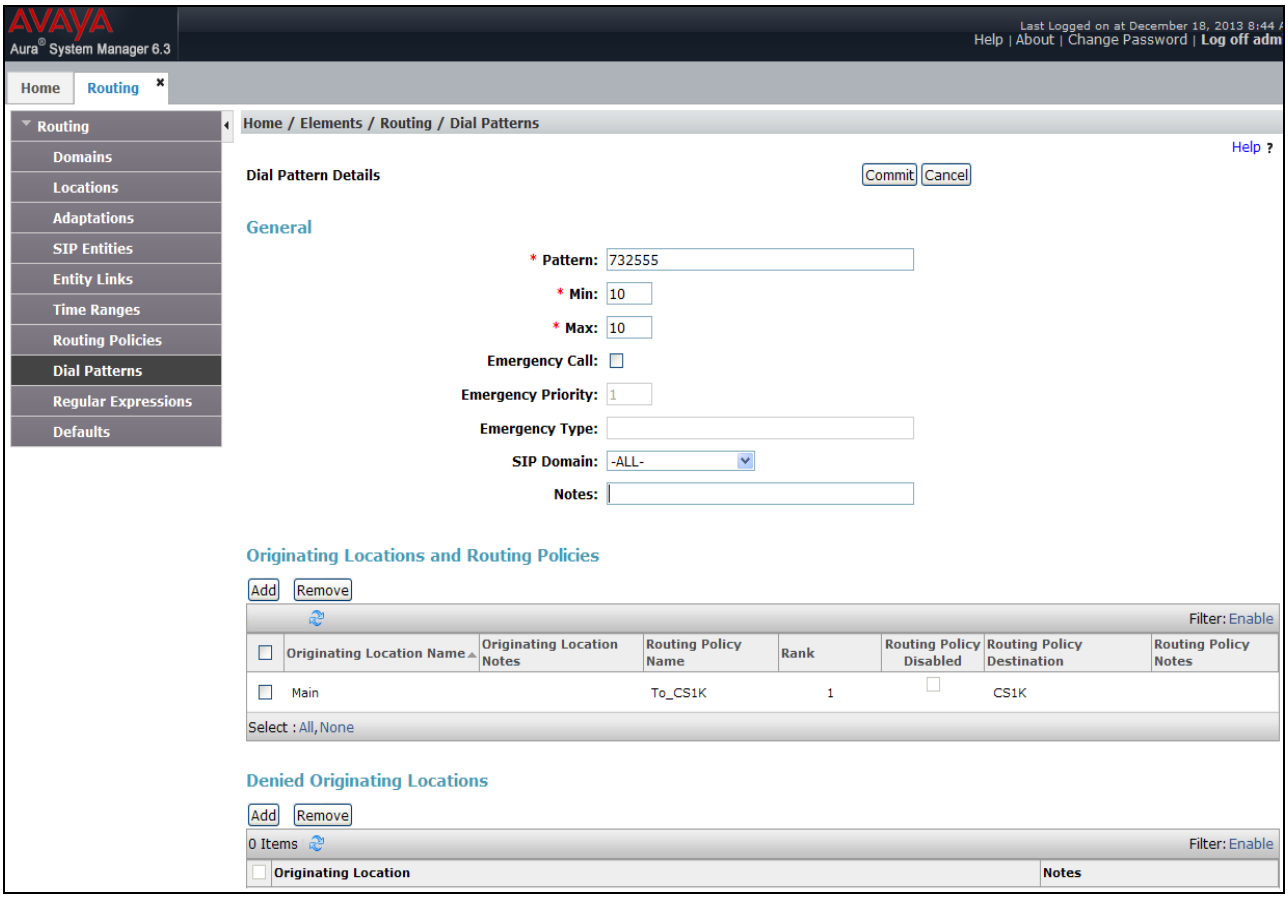
- In the **Originating Location** list, select the location **Main**, (which covers the Avaya SBCE IP address), defined in **Section 6.2.2**.
- In the **Routing Policies** table, select the **To_CS1K** Routing Policy defined in **Section 6.6.1**.
- Click **Select** to save these changes and return to **Dial Pattern Details** page.

Step 4 - Click **Commit** to save.

Step 5 - Repeat **Steps 1-4** as needed for any additional inbound AT&T DNIS number patterns.

Note – No **Denied Originating Locations** were specified.

Note – The AT&T IP Toll Free service is inbound only, so no outbound dial patterns are specified.



7. Avaya Aura® Contact Center

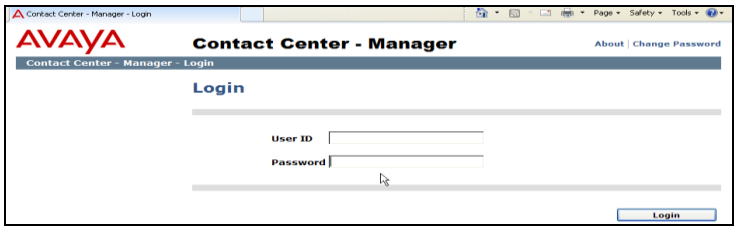
In the reference configuration, Avaya Aura® Contact Center is used to manage Agent functionalities and integrate these functions to the CS1000E.

Note - In the reference configuration, Application Module Link (AML) protocol is used between the CS1000E and Avaya Aura® Contact Center. The provisioning and establishment of the AML connection between Avaya Aura® Contact Center and the CS1000E is assumed to be completed. However, SIP based connections are also supported.

Note – The installation and initial provisioning of Avaya Aura® Contact Center is beyond the scope of this document (see [**11-14**] for more information). Only the Agent provisioning supporting the AT&T IP Toll Free solution testing, is shown below.

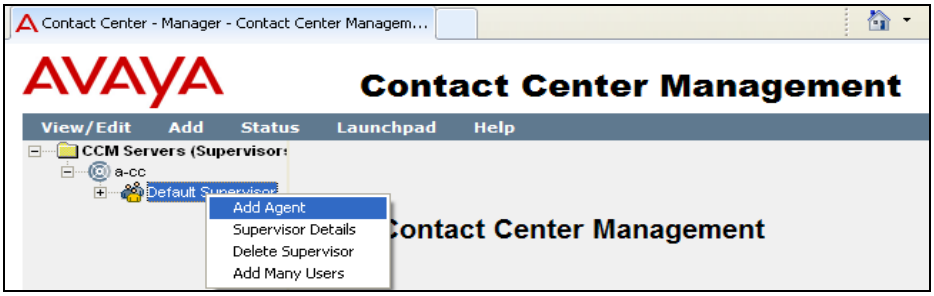
7.1. Create Avaya Aura® Contact Center Agent

Step 1 – Log into the Avaya Aura® Contact Center Manager web interface.

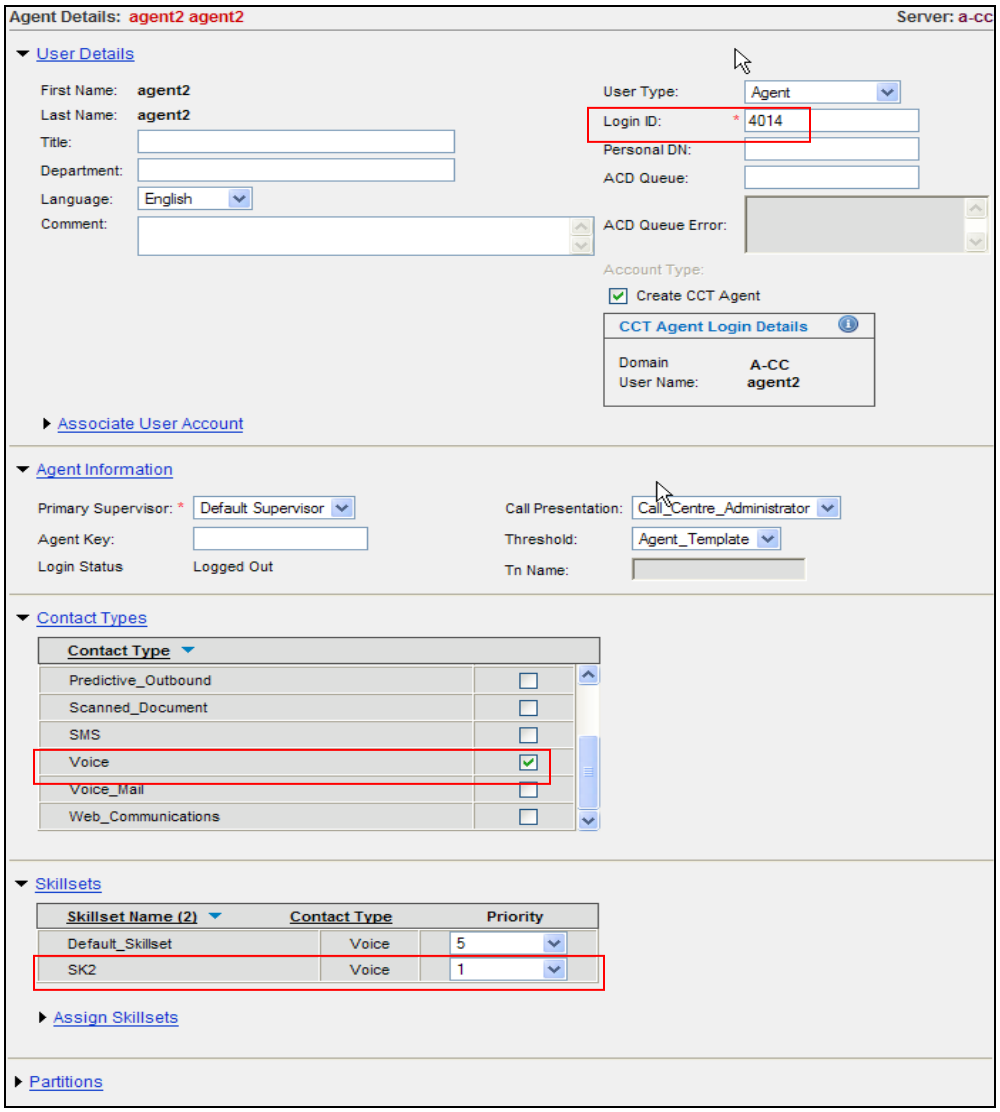


Step 2 – On the **Launchpad** page, select **Contact Center Management**.

Step 3 – In the left hand column, expand the name of the Avaya Aura® Contact Center (e.g., **a-cc**), right click on appropriate supervisor (e.g., **Default Supervisor**), and select **Add Agent**.



Step 4 – On the **Agent Details** page, enter the information as shown in the example below. In the example, **agent2** has a login ID of **4014** (see **Section 5.7**), is a **Voice** Contact, and is assigned as a priority 1 contact for skill set two (**SK2**).



Step 5 – Click **Submit** (not shown). Repeat **Steps 1-5** for additional Agents/Skills.

7.2. Verify Control DN (CDN) and Agent Connection Status

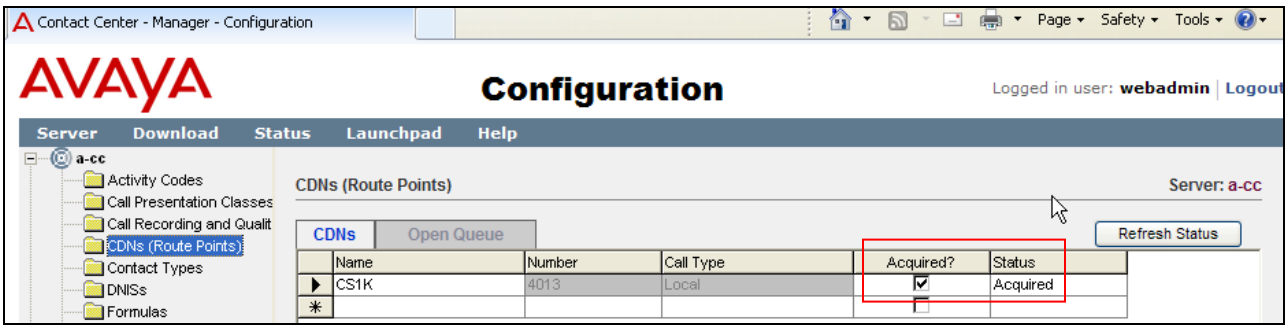
7.2.1. CDN Connection status

The Avaya Aura® Contact Center/CS1000E CDN connection status can be verified as follows.

Step 1 – Connect to **Launchpad** as described in **Section 7.1**.

Step 2 – Select **Configuration**.

Step 3 – From the left hand menu select **CDNs (Route Points)**. The connection provisioned on Avaya Aura® Contact Center to the CS1000E will be displayed. Verify the status is **Acquired**.

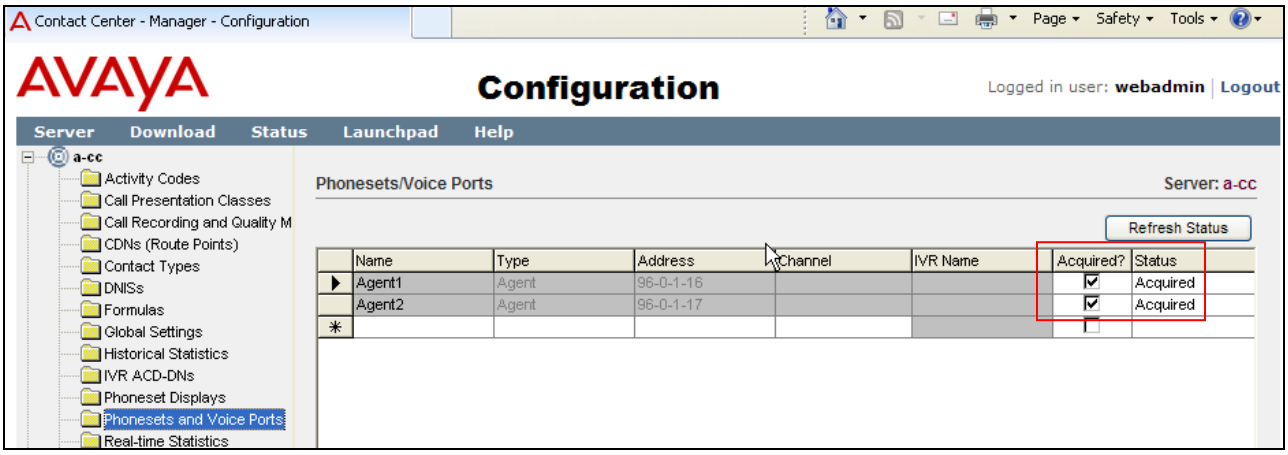


7.2.2. Agent Connection status

Step 1 – Connect to **Launchpad** as described in **Section 7.1**.

Step 2 – Select **Configuration**.

Step 3 – From the left hand menu select **Phonesets and Voice Ports**. The provisioned agents will be displayed. Verify the status is **Acquired**.



8. Configure Avaya Session Border Controller for Enterprise

8.1. Initial Installation/Provisioning

Note: The installation and initial provisioning of the Avaya SBCE is beyond the scope of this document. Refer to **[9-10]** for additional information.

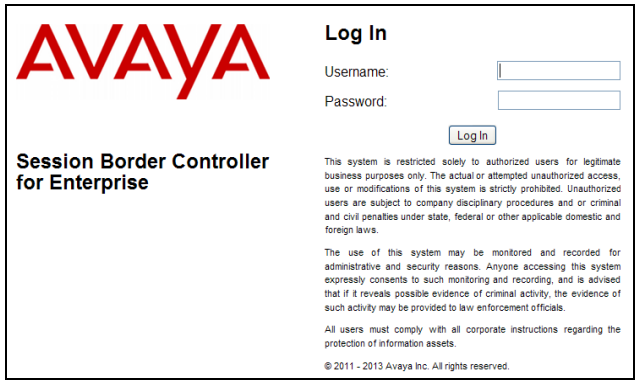
IMPORTANT! – During the Avaya SBCE installation, the Management interface of the Avaya SBCE <u>must</u> be provisioned on a different subnet than either of the Avaya SBCE private and public network interfaces (e.g., A1 and B1). If this is not the case, contact your Avaya representative to get this condition resolved.

In the reference configuration, the Avaya SBCE interface B1 (192.168.64.130) was used for the public interface (toward AT&T), and interface A1 (192.168.67.120) was the private network interface.

8.2. Log into the Avaya SBCE

The follow provisioning is performed via the Avaya SBCE GUI interface.

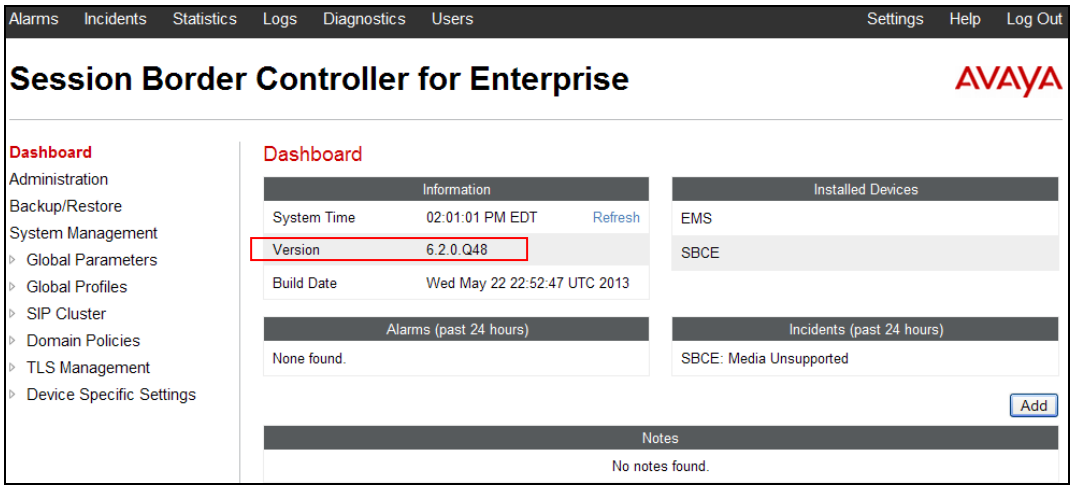
Step 1 - Access the web interface by typing "**https://x.x.x.x**" (where x.x.x.x is the management IP address of the Avaya SBCE).
Step 2 – Enter the appropriate credentials, and click on **Log In**.



The Avaya SBCE Dashboard screen is displayed. All platform navigation is performed from the menu area on the left of the screen. This menu is displayed for all screens.

The platform version used in the reference configuration is displayed in the center of the display (e.g., **6.2.0 Q48**).

Note – See **Section 2.2.1, Item 6**, regarding the Avaya SBCE platform version.

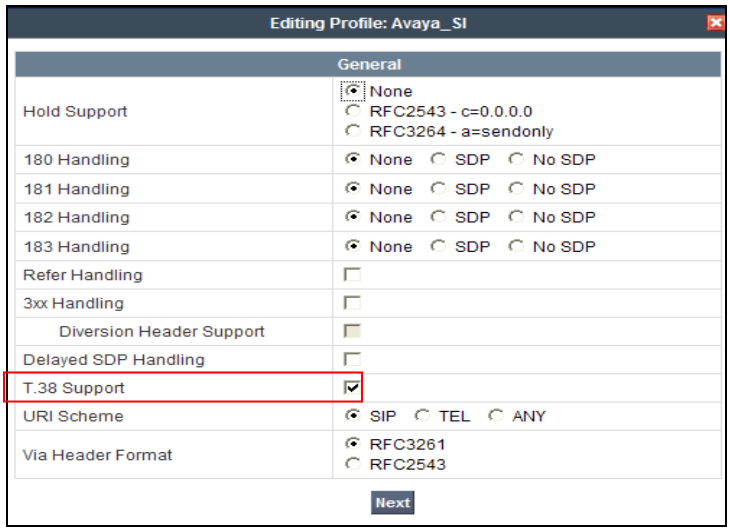


8.3. Global Profiles

8.3.1. Server Interworking – Avaya Side

Step 1 - Select **Global Profiles → Server Interworking** (not shown).
Step 2 - Select the **Add** button (not shown).
Step 3 - Enter a profile name (e.g., "**Avaya_SI**") and click on **Finish**. The new profile name will appear on the profile list.
Step 4 - Select the profile name created above, and then select the **General** Tab (not shown). Scroll down and click on **Edit** (not shown):

- Check **T38 Support →Yes**
- All other options on the General Tab can be left at default
- Select **Next**



Step 5 - Accept default values on all remaining tabs**,** then click **Finish** (not shown).

8.3.2. Server Interworking – AT&T Side

Repeat the steps shown in **Section 8.3.1** to add an Interworking Profile for the connection to AT&T.

Step 1 - On the **General** Tab:

- Enter a profile name**:** (e.g., "**ATT_SI**").
- Check **T38 Support**.
- All other options on the General Tab can be left at default.
- Select **Next**.

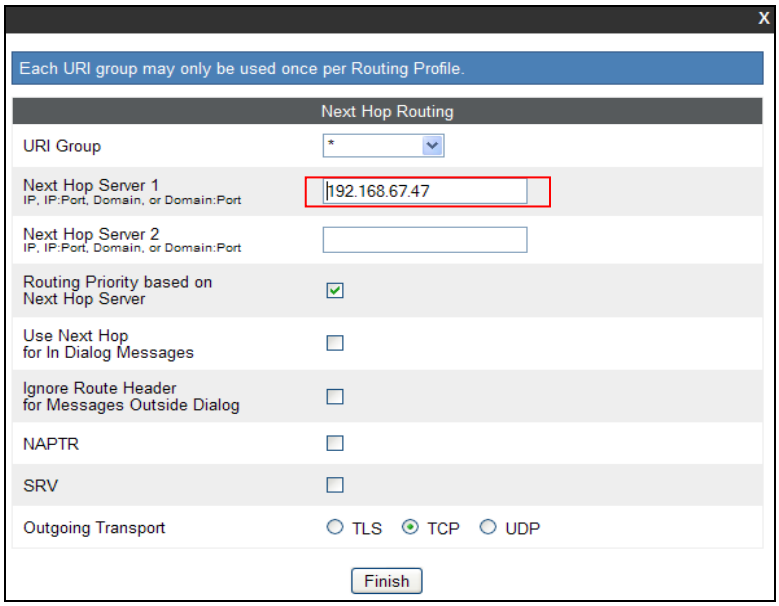
Step 2 - Accept default values on all remaining tabs**,** then click **Finish** (not shown).

8.3.3. Routing – Avaya Side

Step 1 - Select **Global Profiles → Routing** from the menu on the left-hand side (not shown).
Step 2 - Select **Add Profile** (not shown).
Step 3 - Enter Profile Name: (e.g., "**To_SM_RP**"**)**.
Step 4 - Click **Next** and enter the following:

- Leave **URI Group** with the default ***** value.
- Set **Next Hop Server 1:** to **192.168.67.47** (Session Manager IP address).
- Select **Routing Priority Based on Next Hop Server**.
- Set **Outgoing Transport**: to **TCP**.

Step 5 - Click **Finish**.



8.3.4. Routing – AT&T Side

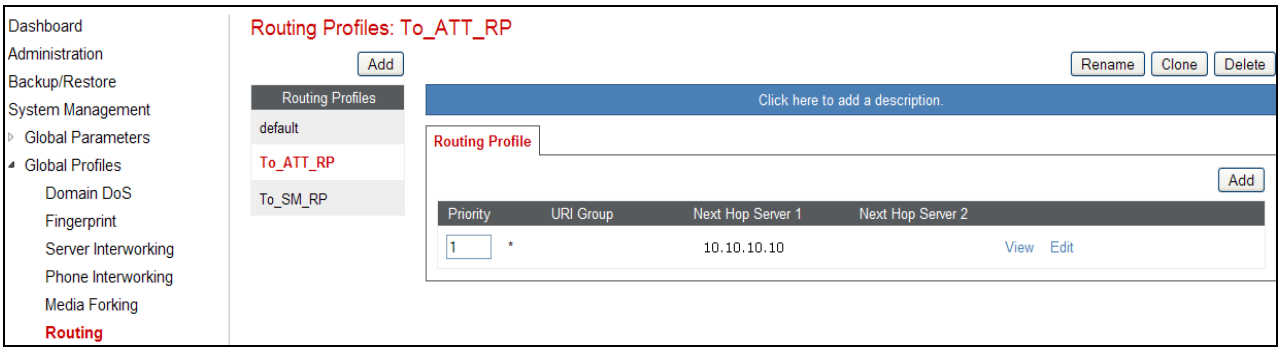
Repeat the steps in **Section 8.3.3** to add a Routing Profile for the AT&T primary Border Element.

Note – See **Appendix 1** for provisioning a route to the AT&T IP Toll Free service secondary Border Element, if applicable.

Step 1 - Select **Add Profile**.
Step 2 - Enter Profile Name: (e.g., "**To_ATT_RP**").
Step 3 - Click **Next**, then following the procedures shown in **Section 8.3.3**, enter the following:

- Set **Next Hop Server 1:** to **10.10.10.10**[4] (AT&T Border Element IP address).
- Select **Routing Priority Based on Next Hop Server**.
- Set **Outgoing Transport**: to **UDP**.

Step 4 - Click **Finish**. The completed form is shown below.



8.3.5. Server Configuration –Session Manager

Step 1 - Select **Global Profiles → Server Configuration** from the menu on the left-hand side (not shown).
Step 2 - Select **Add Profile** and the **Profile Name** window will open (not shown). Enter a Profile Name (e.g., "**SM_SC**") and select **Next**.
Step 3 - The **Add Server Configuration Profile - General** window will open (not shown). Enter the following:

- Set **Server Type:** to **Call Server**.
- Set **IP Address:** to **192.168.67.47** (Session Manager IP Address)**.**
- For **Supported Transports**: check **UDP** and **TCP**.
- Set **TCP Port:** to **5060**.
- Set **UDP Port:** to **5060**.
- Select **Next**.

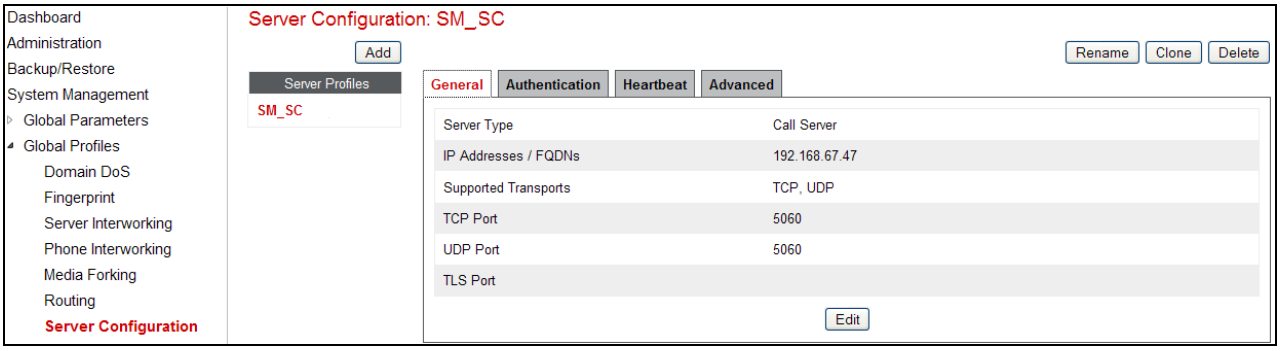
Step 4 - The **Authentication** window will open (not shown). Select **Next** to accept default values.
Step 5 - The **Heartbeat** window will open (not shown). Select **Next** to accept remaining default values.

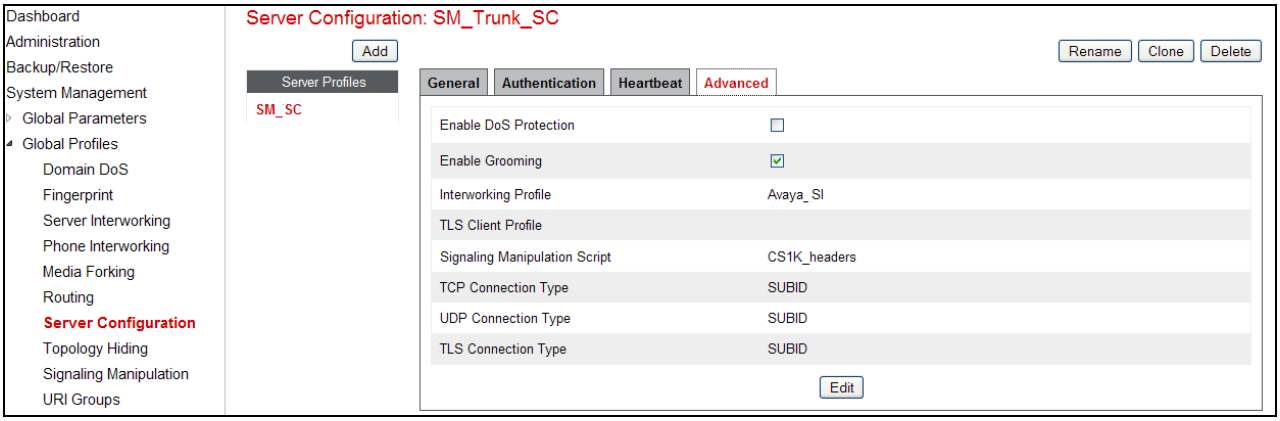
[4] See the note in **Section 3.1** regarding this address

Step 6 - The **Advanced** window will open.

- Select **Enable Grooming**.
- For **Interworking Profile** select **Avaya_SI** created in **Section 8.3.1**.
- For the **Signaling Manipulation Script** select the **CS1K_headers** script defined in **Section 8.3.9**.
- Select **Finish**, accepting remaining default values.

The following screen shots show the completed **General** and **Advanced** tabs.





8.3.6. Server Configuration –AT&T Primary Border Element

Note – See **Appendix 1** for configuration of a Secondary AT&T IP Toll Free order Element, if applicable.

Repeat the steps in **Section 8.3.5** to create a Server Configuration for the connection to the AT&T primary Border Element, using the following entries:

Step 1 - In the **Profile Name** window enter a Profile Name (e.g., "**ATT_Primary_SC**") and select **Next**.
Step 2 – In the **Add Server Configuration Profile - General** window for **Server Type:** select **Trunk Server.**

- Enter **IP Address: 10.10.10.10**[5] (AT&T IP Toll Free primary border element).
- For **Supported Transports**: check **UDP**
- For **UDP Port:** enter **5060**
- Select **Next**

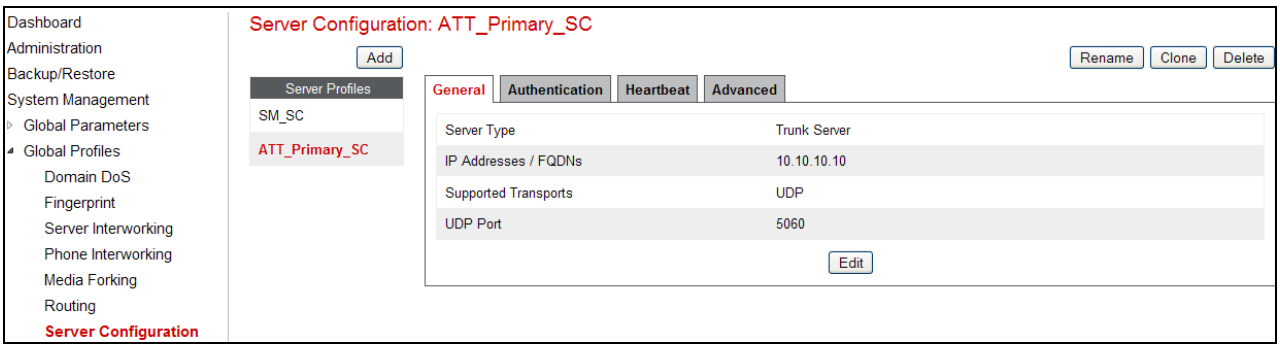
Step 3 - Accept default values for the **Add Server Configuration Profile - Authentication** and **Heartbeat** windows (not shown).

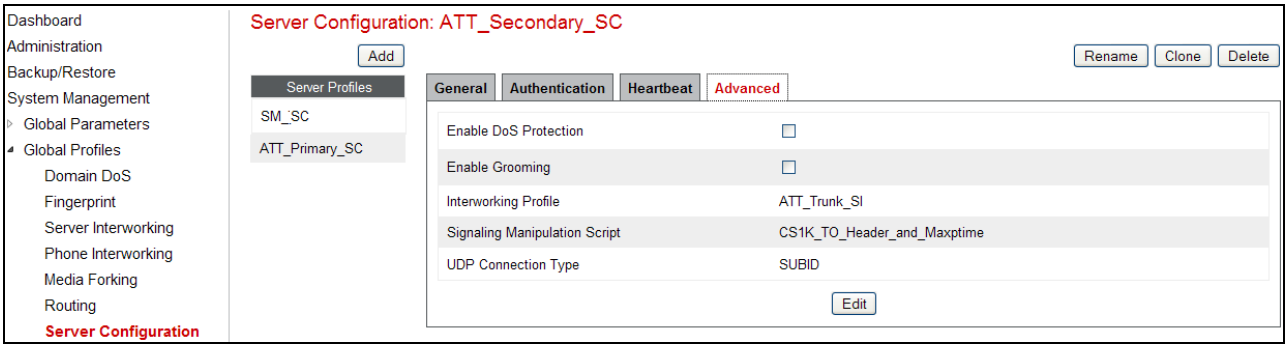
Step 4 – The **Add Server Configuration Profile - Advanced** window will open.

- Select **ATT_SI** for **Interworking Profile** (created in **Section 8.3.2**).
- For the **Signaling Manipulation Script** select the **CS1K_TO_Header_and_Maxptime** script that was defined in **Section 8.3.9**.

Step 5 - Select **Finish**.

The following screens show the completed **General** and **Advanced** tabs.





8.3.7. Topology Hiding – Avaya Side

Step 1 - Select **Global Profiles → Topology Hiding** from the menu on the left-hand side (not shown).

Step 2 - Click **default** profile and select **Clone Profile**.

Step 3 - Enter Profile Name: (e.g., "**Avaya_TH**"). Enter the following:

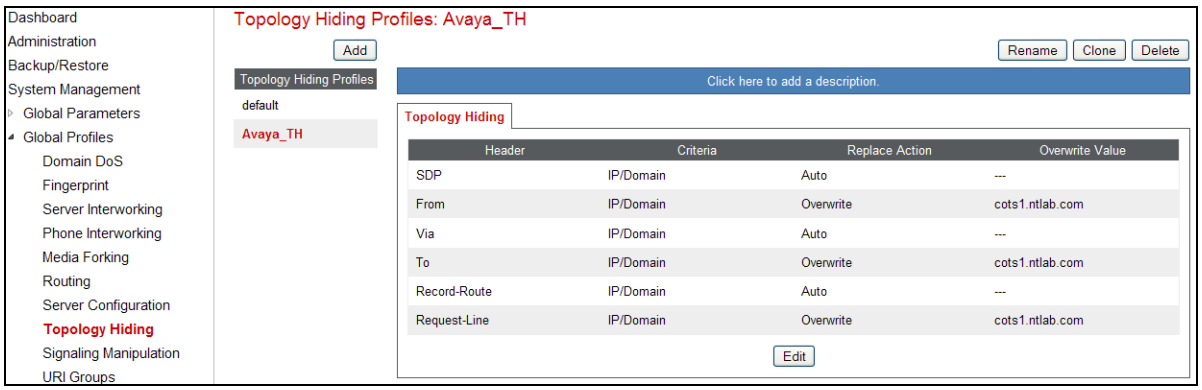
- For the Header **To**,
  - In the **Criteria** column select **IP/Domain**

[5] See the note in **Section 3.1** regarding this address

- In the **Replace Action** column select**: Overwrite**
- In the **Overwrite Value** column**: cots1.ntlab.com**

- Repeat for the Header **From**
- Repeat for the Header **Request Line**

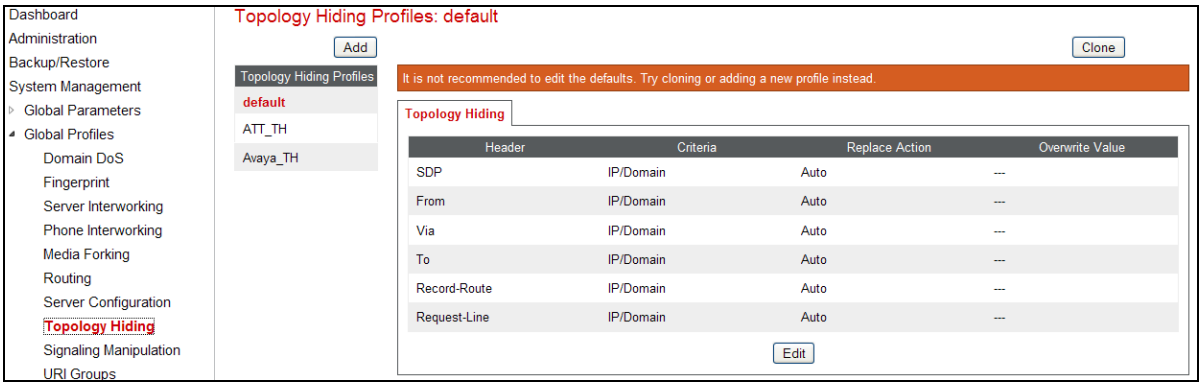
Step 4 - Click **Finish** (not shown).



8.3.8. Topology Hiding – AT&T Side

Create a **Topology Hiding Profile** for the connection to AT&T, by repeating the steps in **Section 8.3.7** with the following changes:

- Enter **Profile Name**: (e.g., "**ATT_TH**").
- Use the default **Replace Action** setting of **Auto**.
- Click **Finish**.



8.3.9. Signaling Manipulation

The Avaya SBCE can manipulate inbound and outbound SIP headers. In the reference configuration, two signaling manipulation scripts are used; **CS1K_TO_Header_and_Maxptime** and **CS1K_headers**.

Note – Use of the Signaling Manipulation scripts require higher processing requirements on the Avaya SBCE. Therefore, this method of header manipulation should only be used in cases where the use of Signaling Rules (**Section 8.4.3**) does not meet the desired result. Refer to **[10]** for information on the Avaya SBCE scripting language.

8.3.9.1 CS1K_TO_Header_and_Maxptime

Calls can be made directly into Call Pilot® (e.g., to check/retrieve messages, or access an Auto Attendant). Call Pilot® checks the contents of the INVITE TO header as part of admission control. However while AT&T sends the unique DNIS number in the INVITE R-URI, it will send the customers Billing Number in the TO header of all INVITEs. Therefore the Avaya SBCE is used to copy the contents of the R-URI, into the TO header, so that Call Pilot® can apply admission control successfully (see **Section 2.2.1, Item 5)**. The script then returns the TO header back to the Billing number in messages sent back to AT&T. See **Section 5.10** for additional provisioning required in Call Pilot® to accept call to check/retrieve messages.

In addition, as described in **Section 2.2.1, Item 1,** AT&T sends INVITEs with the SIP parameter *maxptime:30*. In response, Avaya CS1000E will send *ptime:10* for any UNIStim or digital stations. The Avaya SBCE is used to change the *maxptime:30* parameter to *ptime:30*, thereby making Avaya CS1000E respond with *ptime:30* as required.

Step 1 - Select **Global Profiles → Signaling Manipulation** from the menu on the left-hand side of the screen (not shown).

Step 2 - Click **Add Script** (not shown) and the script editor window will open.
Enter a name for the script in the **Title** box (e.g., "**CS1K_TO_Header_and_Maxptime**"). Enter the script body as shown below.

Step 3 - Click on **Save**. The script editor will test for any errors, and the Edit window will close. This script is applied to the AT&T Server Configuration in **Section 8.3.6, Step 4**.

The completed script is shown below.

```
// Replace inbound TO header billing number with RURI DNIS number for CS1K.
Apply to AT&T side.

within session "ALL"
{
act on request where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING"
  {
// Replace Billing number in "TO" with "REQUEST-LINE" number

  %touser = %HEADERS["To"][1].URI.USER;

  %HEADERS["To"][1].URI.USER = %HEADERS["Request_Line"][1].URI.USER;
  }
}
// Return ?TO? header to original form
within session "ALL"
{
act on response where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
  {
  %HEADERS["To"][1].URI.USER = %touser;
  }
}

// Replace maxptime:30 with ptime:30 in calls to CS1K.

within session "ALL"
{
  act on request where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING"
  {
    %BODY[1].regex_replace( "a=maxptime:30","a=ptime:30");
  }
}
```

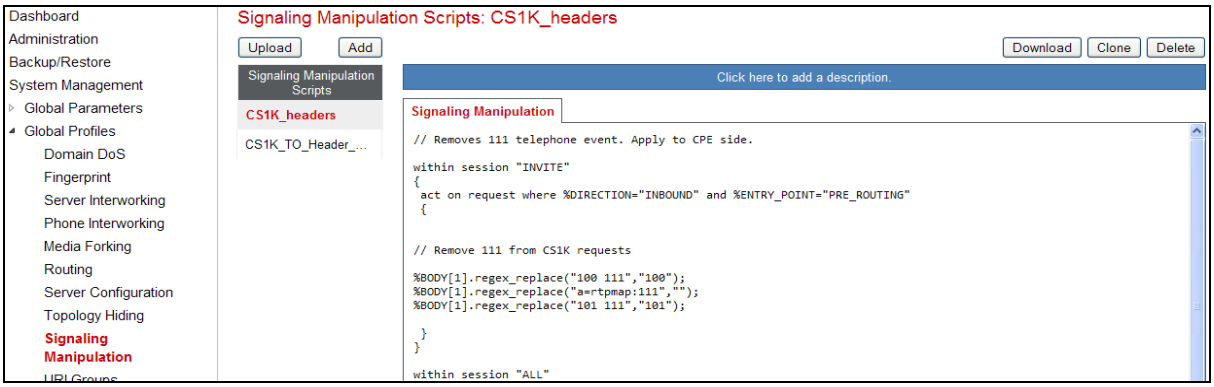
8.3.9.2 CS1K_headers

As described in **Section 2.2.1, Item 5**, the Avaya CS1000E inserts a telephone event type of 111 which AT&T does not support. This value is removed via the following script. In addition, in some call scenarios the Avaya CS1000E may insert a leading + in the calling/called number fields. This is also not required by AT&T, and is removed.

- Select **Global Profiles → Signaling Manipulation** from the menu on the left-hand side of the screen (not shown).
- Click **Add Script** (not shown) and the script editor window will open.
- Enter a name for the script in the **Title** box (e.g., "**CS1K_headers**"). Enter the script body as shown below.
- Click on **Save**. The script editor will test for any errors, and the Edit window will close. This script is applied to the Avaya Server Configuration in **Section 8.3.5**, **Step 6**.

```
// Removes 111 telephone event. Apply to CPE side.
// Remove 111 from CS1K requests

within session "INVITE"
{
 act on request where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING"
 {
%BODY[1].regex_replace("100 111","100");
%BODY[1].regex_replace("a=rtpmap:111","");
%BODY[1].regex_replace("101 111","101");
 }
}

// Remove 111 from CS1K responses

within session "ALL"
{
 act on response where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING"
 {
%BODY[1].regex_replace("100 111","100");
%BODY[1].regex_replace("a=rtpmap:111","");
%BODY[1].regex_replace("101 111","101");
 }
}

// Remove plus sign from From, Contact, and PAI

within session "INVITE"
{
  act on request where %DIRECTION="INBOUND" and %ENTRY_POINT="PRE_ROUTING"
  {
    %HEADERS["Request_Line"][1].regex_replace("\+","");
    %HEADERS["From"][1].regex_replace("\+","");
    %HEADERS["Contact"][1].regex_replace("\+","");
    %HEADERS["P-Asserted-Identity"][1].regex_replace("\+","");
  }
}
```

The screen below shows the completed Signaling Manipulations form.



8.4. Domain Policies

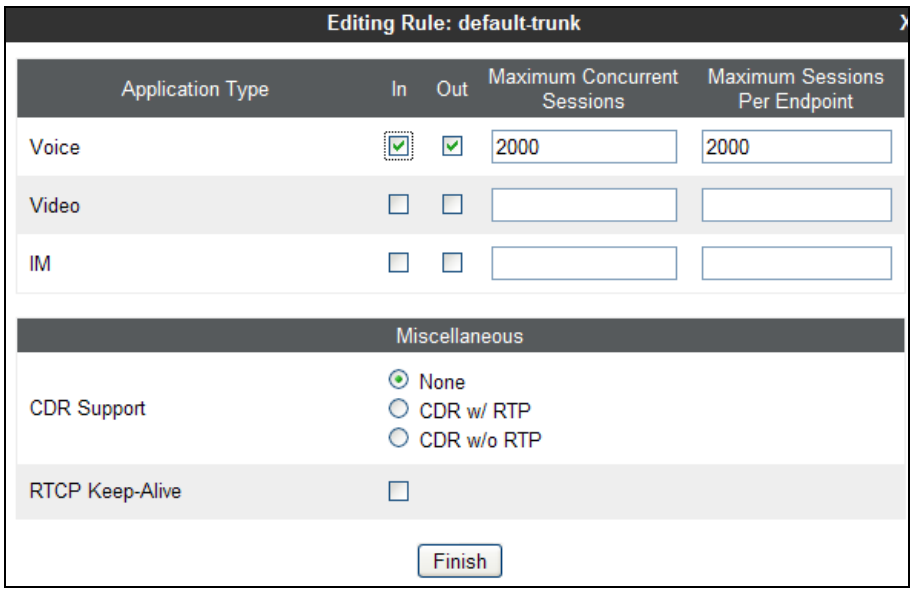
8.4.1. Application Rules

Step 1 - Select **Domain Policies → Application Rules** from the menu on the left-hand side menu (not shown).
Step 2 - Select the **default** Rule
Step 3 - Select **Clone Rule** button

- For **Name:** enter "**SIP_Trunk_AR**"
- Click **Finish**

Step 4 - Highlight the rule **SIP_Trunk_AR** just created, and click the **Edit** button.

- In the **Voice** row:
  - Change the **Maximum Concurrent Sessions** to an appropriate amount (e.g., **2000)**
  - Change the **Maximum Sessions per Endpoint** to an appropriate amount (e.g.,**2000)**
  - Set **CDR Support** to **None**.
  - Click on **Finish**.



8.4.2. Media Rules

8.4.2.1 Avaya Media Rule

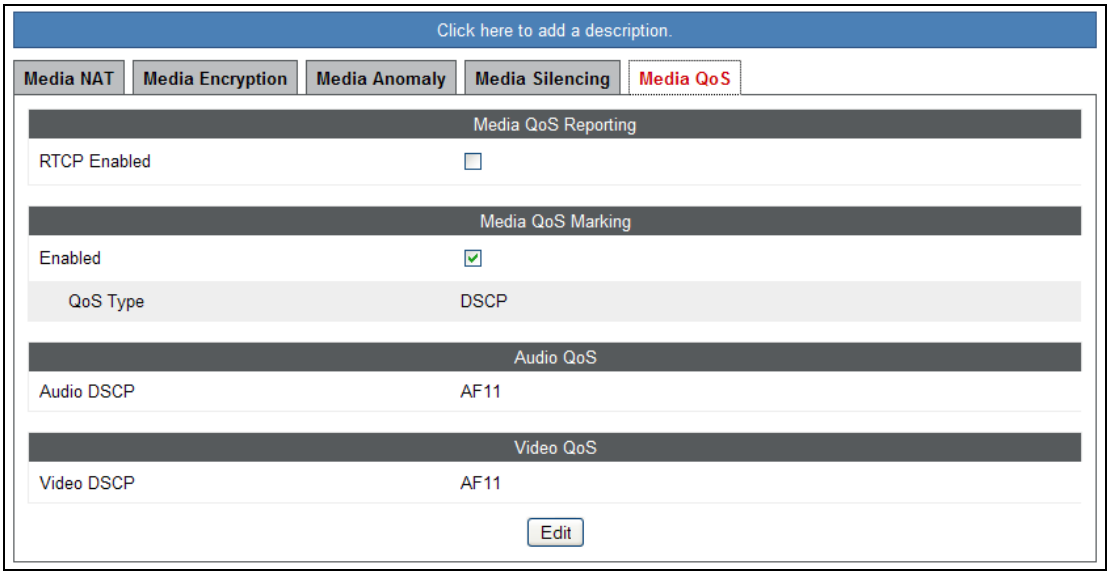
Step 1 - Select **Domain Policies → Media Rules** from the menu on the left-hand side menu (not shown).
Step 2 - From the Media Rules menu, select the **default-low-med** rule
Step 3 - Select **Clone Rule** button

- Name**: Avaya_trunk_low_med**
- Click **Finish**

Step 4 - Highlight the **Avaya_trunk_low_med** rule just created, select the **Media QOS** tab, and click the **Edit** button.

- Check the **Media QOS Marking - Enabled**
- Select the **DSCP** box
- **Audio:** Select **AF11** from the drop-down
- **Video**: Select **AF11** from the drop-down

Step 5 - Click **Finish** (not shown).



8.4.2.2 AT&T Media Rule

Step 1 – Repeat the steps in **Section 8.4.2.1** with the following changes:

- Name**: ATT _low_med**

Step 2 - Click **Finish** (not shown).

8.4.3. Signaling Rules

As described in **Section 2.2.1, Item 2**, the Avaya SBCE is used to help reduce packet size by removing SIP headers not required by AT&T.

8.4.3.1 Avaya - Requests

Step 1 - Select **Domain Policies → Signaling Rules** from the menu on the left-hand side menu (not shown).

Step 2 - From the Signaling Rules menu, select the **default** rule.

Step 3 - Select **Clone Rule** button

- Enter a name**: CS1K_SR_with_SM**
- Click **Finish**

Step 4 - Select the **CS1K_SR_with_SM** rule and do the following:

- Select the **Request Headers** tab (not shown), and select the **Add In Header Control** button (not shown).
- Click the **Edit** button and the **Edit Header Control** window will open (not shown).
- Check the **Proprietary Request Header** box.

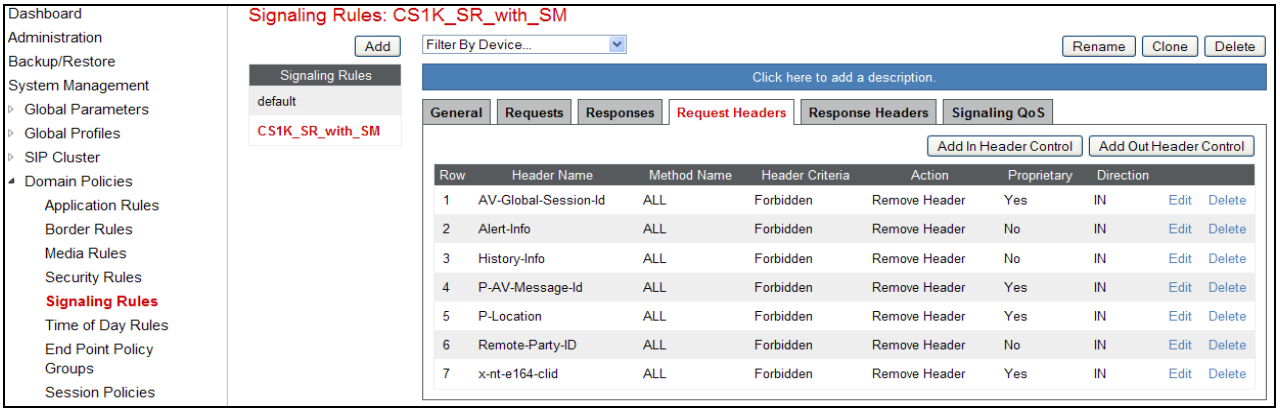
- From the **Header Name** menu select **P-Location**.
- From the **Method Name** menu select **All**.
- For **Header Criteria** select **Forbidden**.
- From the **Presence Action** menu select **Remove Header**.
- Click **Finish**

Step 5 - Repeat **Step 4** to create a rule to remove the **P-AV-Message-ID**, **P-Location**, and **x-nt.E164-clid** proprietary headers.

Step 6 - Repeat **Step 4** to remove the **Alert-Info**, **History-Info**, and **Remote-Party-ID** non proprietary headers.

- Do *not* check the **Proprietary Request Header** box.

The completed form is shown below. Note that all the entries in the **Direction** column says **In**.



8.4.3.2 Avaya - Responses

Following the steps shown in **Section 8.4.3.1**, Response Signaling Rules are defined to remove **AV-Global-Session-ID**, **History-Info**, **P-AV-Message-ID**, **Remote-Party-ID**, and **P-Location** headers for both **1xx** and **2xx** responses.

Step 1 - Highlight the **CS1K_SR_with_SM** rule created in **Section 8.4.3.1** and enter the following to remove the **AV-Global-Session-ID** proprietary header from **1XX** responses.

- Select the **Response Headers** tab (not shown).
- Click the **Edit** button and the **Edit Header Control** window will open.
- Check the **Proprietary Request Header** box.
- From the **Header Name** menu enter **AV-Global-Session-ID**.
- From the **Response Code** menu select **1xx**.
- From the **Method Name** menu select **All**.
- For **Header Criteria** select **Forbidden**.
- From the **Presence Action** menu select **Remove Header**.
- Click **Finish**

Step 2 - Repeat **Step 1** to create rules to remove the **P-AV-Message-ID**, **Remote-Party-ID**, and **P-Location** proprietary headers for **1xx** responses.

Step 3 - Repeat **Step 2** to create rules to remove the **History-Info and Remote-Party-ID** *non-proprietary* headers for **1xx** responses.

- Do *not* check the **Proprietary Request Header** box.

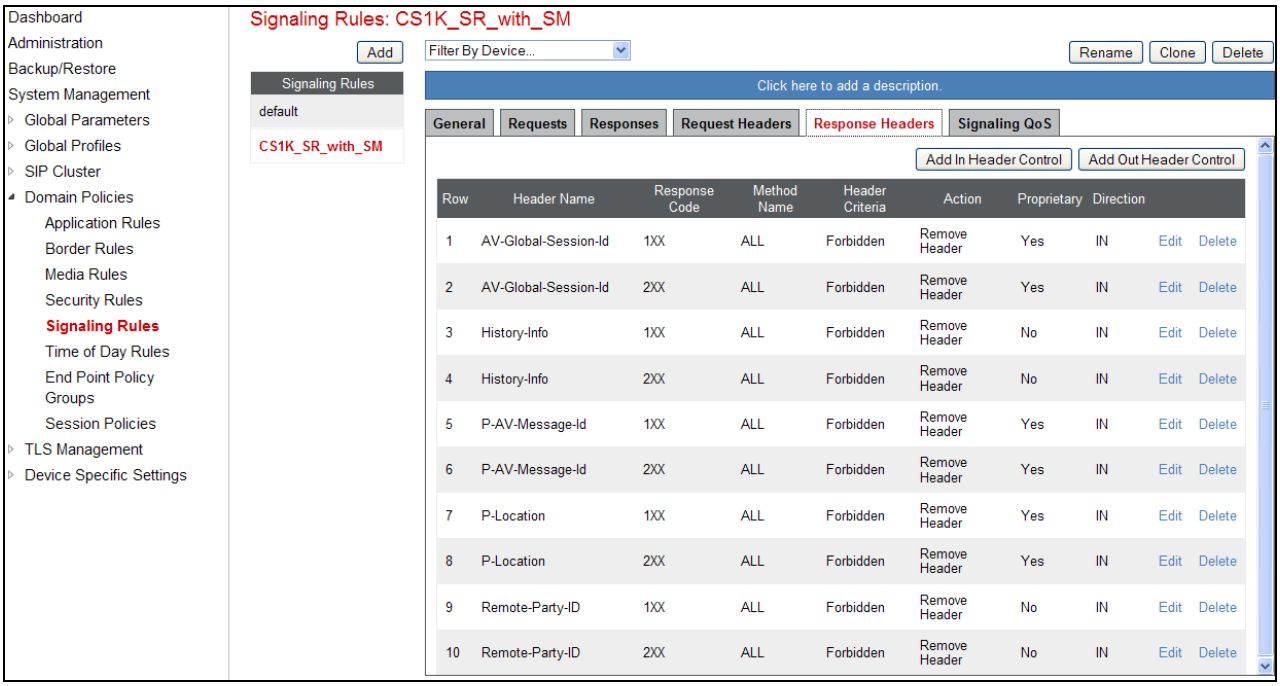
Step 4 - Repeat **Step 1** to create rules to remove **AV-Global-Session-ID**, **P-AV-Message-ID**, and **P-Location** proprietary headers for **2xx** responses

- From the **Response Code** menu select **2xx**.

Step 5 - Repeat **Step 4** to create rules to remove **History-Info**, and **Remote-Party-ID** non-proprietary headers for **2xx** responses

- Do *not* check the **Proprietary Request Header** box.

The completed form is shown below. Note that all the entries in the **Direction** column says **In**.

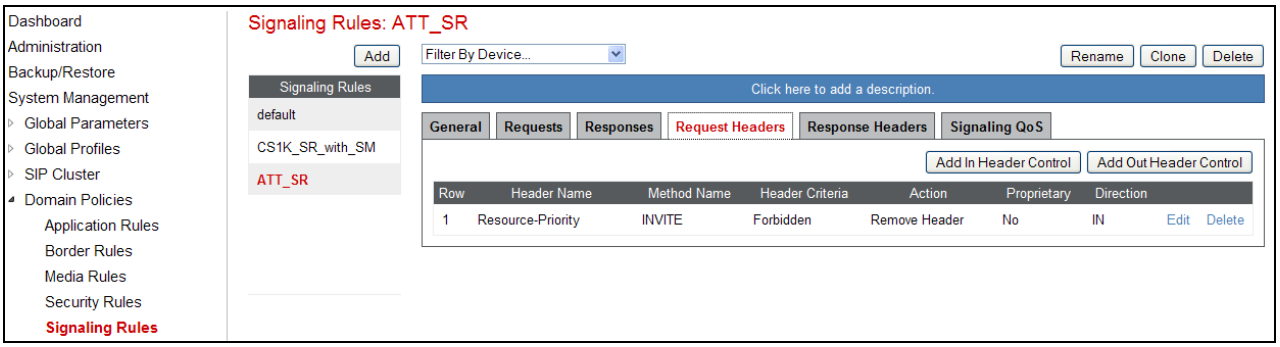


8.4.3.3 AT&T – Requests

Step 1 – Follow the steps in **Section 8.4.3.1**, and create a Request rule called **ATT_SR**.

Step 2 - Select the **ATT_SR** rule and do the following:

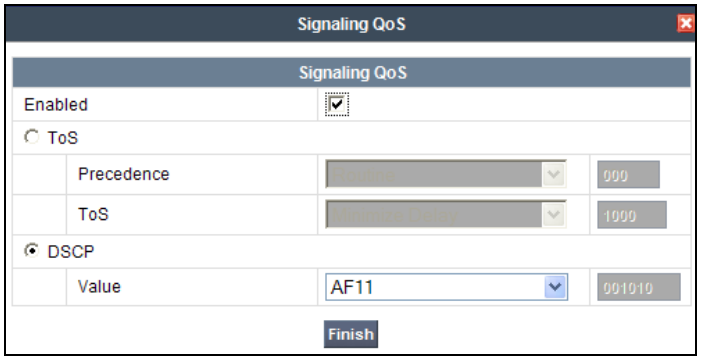
- Select the **Request Headers** tab (not shown), and select the **Add In Header Control** button (not shown).
- Click the **Edit** button and the **Edit Header Control** window will open (not shown).
- Do *not* check the **Proprietary Request Header** box.
- From the **Header Name** menu select **Resource-Priority**.
- From the **Method Name** menu select **Invite**.
- For **Header Criteria** select **Forbidden**.
- From the **Presence Action** menu select **Remove Header**.
- Click **Finish**



8.4.3.4 Avaya – Signaling QOS

Step 1 - Highlight the **CS1K_SR_with_SM** rule created in **Section 8.4.3.1** and enter the following:

- Select the **Signaling QOS** tab (not shown).
- Click the **Edit** button and the **Signaling QOS** window will open.
- Select the **Enabled** option.
- Select **DCSP**.
- Select **Value** = **AF11**.
- Click **Finish**.



8.4.3.5 AT&T – Signaling QOS

Step 1 - Highlight the **ATT_SR** rule created in **Section 8.4.3.3** and repeat the procedure in **Section 8.4.3.4**.

8.4.4. Endpoint Policy Groups – Avaya

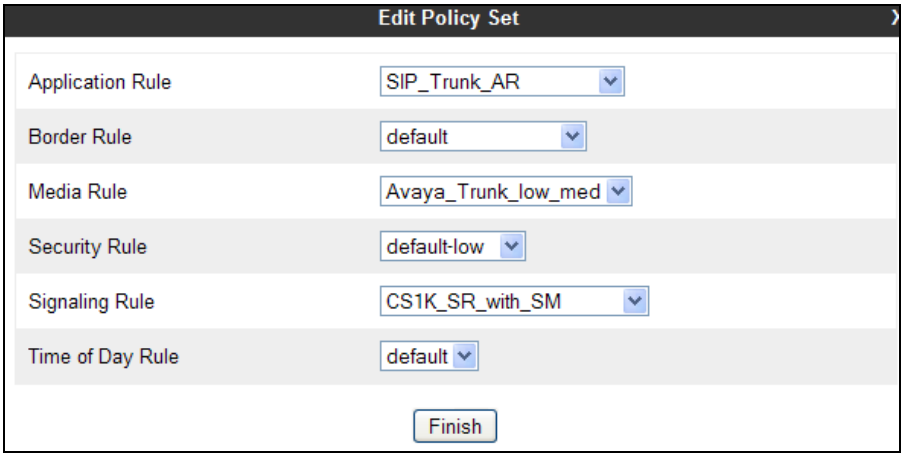
Step 1 - Select **Domain Policies → End Point Policy Groups** from the menu on the left-hand side (not shown).

Step 2 - Select **Add** (not shown).

- For **Name:** enter "**Avaya_default_low_PG**", then click **Next**.
- For **Application Rule:** select **SIP_Trunk_AR** (see **Section 8.4.1**).
- For **Border Rule:** select **default**.
- For **Media Rule:** select **Avaya_Trunk_low_med** (see **Section 8.4.2**).
- For **Security Rule:** select **default-low**.

- For **Signaling Rule:** select **CS1K_SR_with_SM** (see **Section 8.4.3**).
- For **Time of Day:** select **default**.

Step 3 - Select **Finish**.



8.4.5. Endpoint Policy Groups – AT&T

Step 1 – Repeat the steps in **Section 8.4.4** with the following setting changes:

- For **Name:** enter "**ATT_default_low_PG**"
- For **Signaling Rule:** select **ATT_SR** (**see Section 8.4.3**).

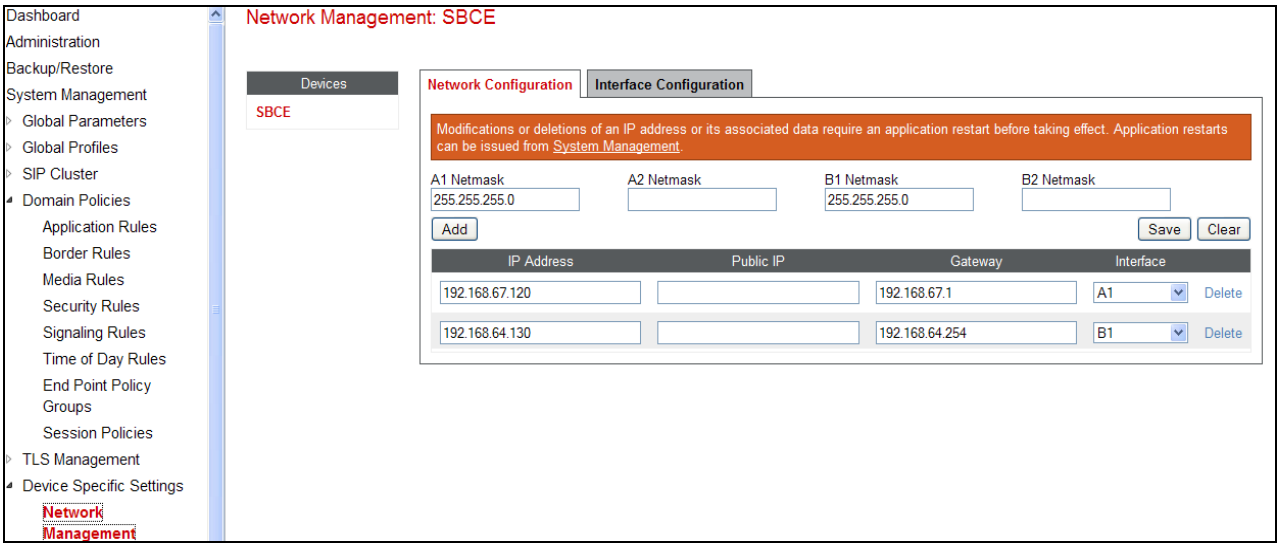
Step 2 - Select **Finish** (not shown).

8.5. Device Specific Settings

8.5.1. Network Management

Step 1 - Select **Device Specific Settings → Network Management** from the menu on the left-hand side.

Step 2 – Select the **Network Configuration** tab. The network interfaces were provisioned during installation. However if these values need to be modified, do so via this tab. In addition, the provisioned interfaces may be enabled/disabled via the **Interface Configuration** tab.



8.5.2. Media Interface

AT&T requires customers to use RTP ports in the range of 16384 – 32767. Both inside and outside ports have been changed but only the outside is recommended by AT&T.

Step 1 - Select **Device Specific Settings → Media Interface** from the menu on the left-hand side, click on **Add**, and enter the following:

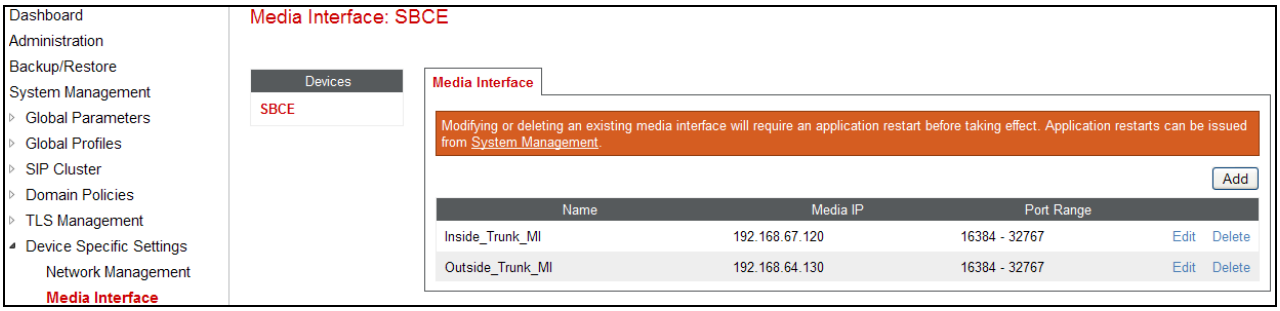
- For **Name:** enter "**Inside_Trunk_MI**"
- For **Media IP:** enter **192.168.67.120** (Avaya SBCE internal address toward Session Manager).
- For **Port Range:** enter **16384 - 32767**

Step 2 - Click **Finish** (not shown)

Step 3 – Repeat **Step 1** with the following changes:

- For **Name:** enter "**Outside_Trunk_MI**"
- For **Media IP:** enter **192.168.64.130** (Avaya SBCE external address toward AT&T)

Step 4 - Click **Finish** (not shown)



8.5.3. Signaling Interface

Step 1 - Select **Device Specific Settings → Signaling Interface** from the menu on the left-hand side.

Step 2 - Select **Add** , and enter the following:

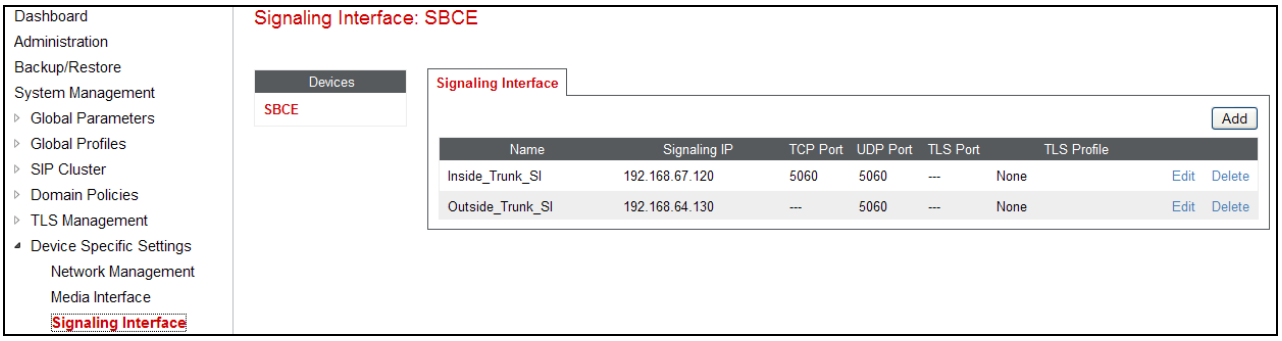
- For **Name:** enter "**Inside_Trunk_SI**"
- For **Media IP:** enter **192.168.67.120** (Avaya SBCE internal address toward Session Manager)
- For **TCP Port:** enter **5060**
- For **UDP Port:** enter **5060**

Step 3 - Click **Finish** (not shown).

Step 4 – Repeat Step 2 with the following changes:

- For **Name:** enter "**Outside_Trunk_SI**"
- For **Media IP:** enter **192.168.64.130** (Avaya SBCE external address toward AT&T).
- For **UDP Port:** enter **5060**

Step 3 - Click **Finish** (not shown).



8.5.4. Endpoint Flows – Avaya (Session Manager)

Step 1 - Select **Device Specific Settings → Endpoint Flows** from the menu on the left-hand side.

Step 2 - Select the **Server Flows** tab

Step 3 - Select **Add**, and enter the following:

- For **Name:** enter "**Avaya_Trunk**"
- For **Server Configuration**: select **SM_SC** (see **Section 8.3.5**)
- For **URI Group:** enter ***** (default)

- For **Transport:** enter ***** (default)
- For **Remote Subnet:** enter ***** (default)
- For **Received Interface**: select **Outside_Trunk_SI** (see **Section 8.5.3**)
- For **Signaling Interface:** select **Inside_Trunk_SI** (see **Section 8.5.3**)
- For **Media Interface**: select **Inside_Trunk_MI** (see **Section 8.5.2**)
- For **End Point Policy Group:** select **Avaya_default_low_PG** (see **Section 8.4.4**)
- For **Routing Profile:** select **To_ATT_RP** (see **Section 8.3.4**)
- For **Topology Hiding Profile:** select **Avaya _TH** (see **Section 8.3.7**)
- For **File Transfer Profile:** enter **None**

Step 4 - Click **Finish** (not shown)

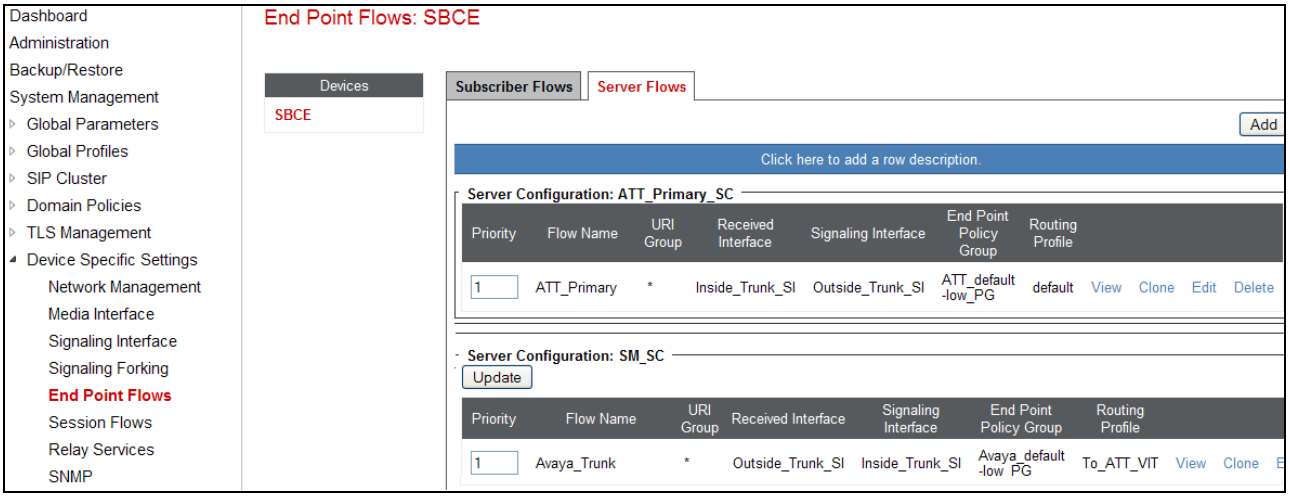
8.5.5. Endpoint Flows – AT&T Primary

Note – See **Appendix 1** for provisioning an Endpoint Flow for the AT&T IPFR-EF service secondary Border Element, if applicable.

Step 1 – Repeat the steps in **Section 8.5.4** with the following changes:

- For **Name:** enter "**ATT_Primary**"
- For **Server Configuration**: select **ATT_Primary_SC** (see **Section 8.3.6**)
- For **Received Interface**: select **Inside_Trunk_SI** (see **Section 8.5.3**)
- For **Signaling Interface:** select **Outside_Trunk_SI** (see **Section 8.5.3**)
- For **Media Interface**: select **Outside_Trunk_MI** (see **Section 8.5.2**)
- For **End Point Policy Group:** select **ATT_default_low_PG** (see **Section 8.4.5**)
- For **Routing Profile:** select **To_SM_RP** (see **Section 8.3.3**)
- For **Topology Hiding Profile:** select **ATT_TH** (see **Section 8.3.8**)

Step 4 - Click **Finish** (not shown)



9. Verification Steps

The following steps may be used to verify the configuration.

9.1. General

[1] Place an inbound call, answer the call, and verify that two-way talk path exists. Verify that the call remains stable for several minutes and disconnect properly.

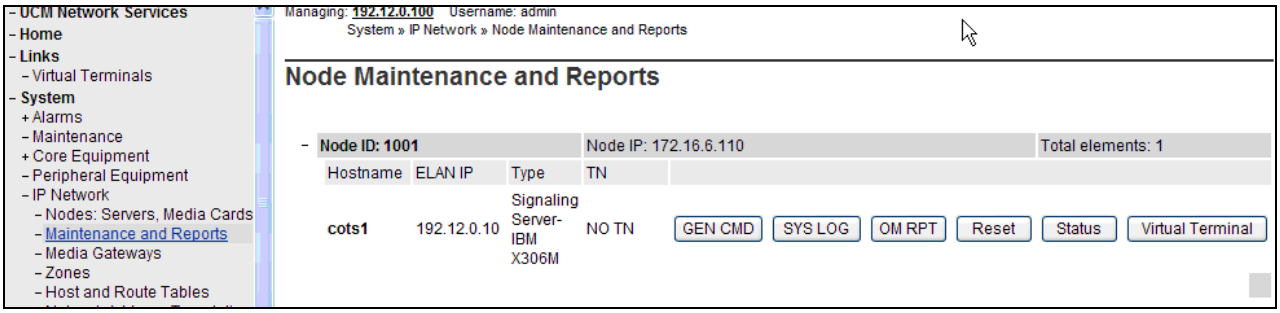
[2] Place an inbound call to an agent or telephone, but do not answer the call. Verify that the call covers to Call Pilot® voicemail. Retrieve the message from Call Pilot®.

9.2. Avaya CS1000E Verifications

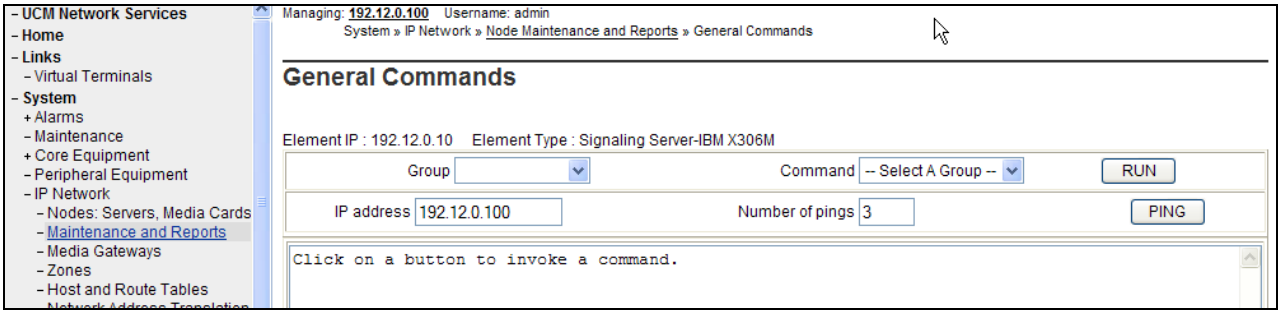
This section illustrates sample verifications that may be performed using the Avaya CS1000E Element Manager GUI.

9.2.1. IP Network Maintenance and Reports Commands

Step 1 - From Element Manager, navigate to **System → IP Network → Maintenance and Reports** as shown below.

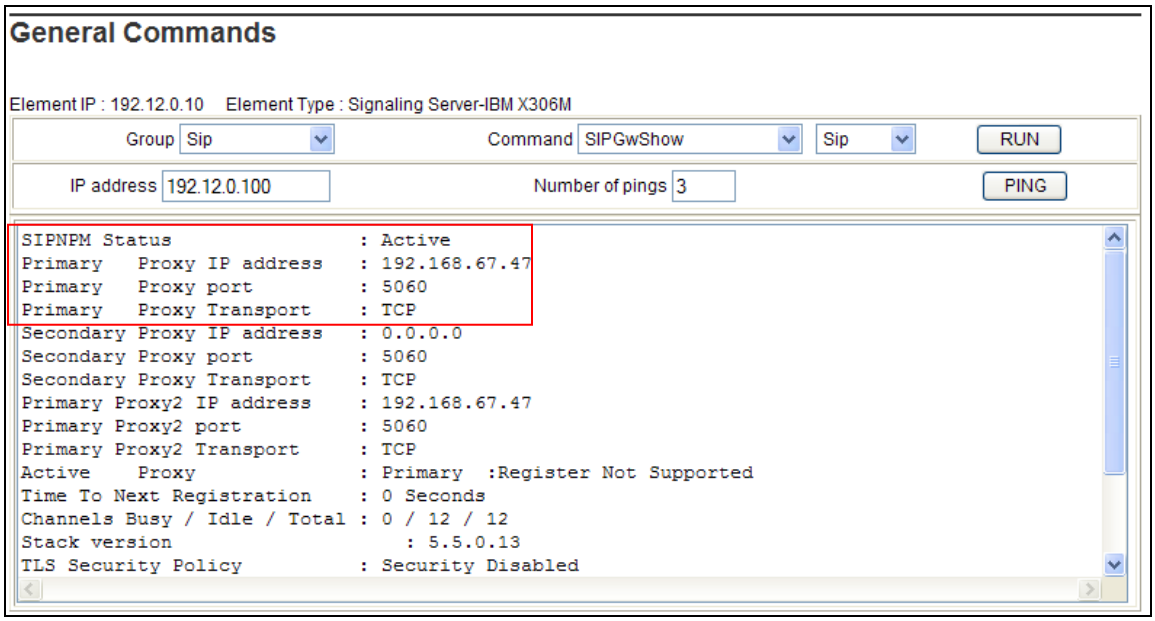


Step 2 - In the resultant screen on the right, click the **Gen CMD** button. The **General Commands** page is displayed as shown below.

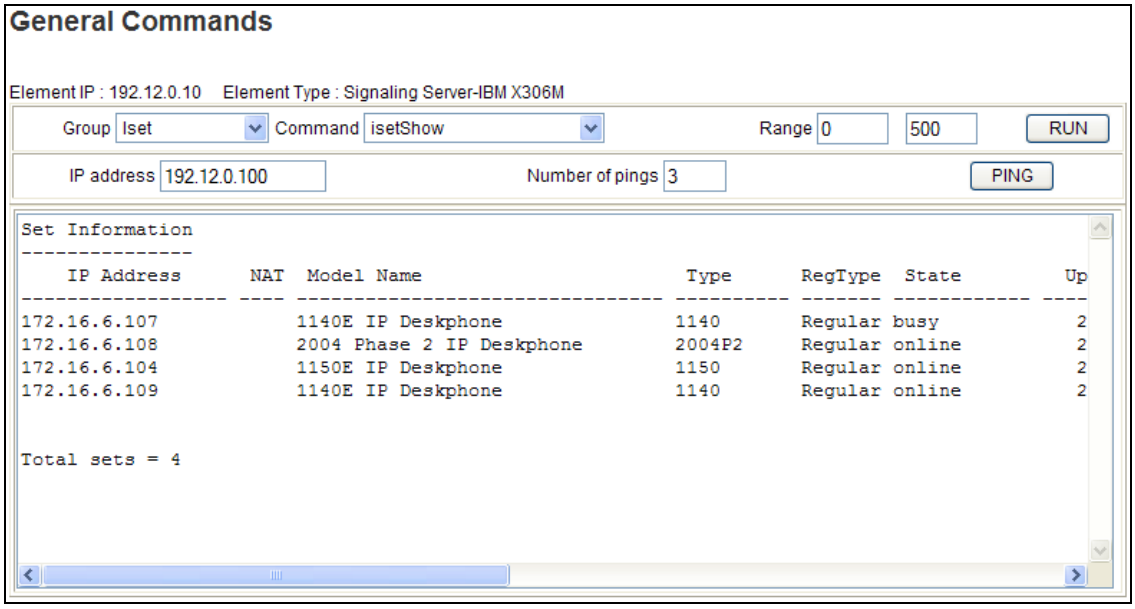


A variety of commands are available by selecting an appropriate **Group** and **Command** from the drop-down menus, and selecting **Run**.

For example, to check the status of the SIP Gateway to Session Manager in the sample configuration, select "**Sip**" from the **Group** menu and "**SIPGwShow**" from the **Command** menu. Click **Run**. The example output below shows that the Session Manager (192.168.67.47, port 5060, TCP) has "SIPNPM Status" Active.

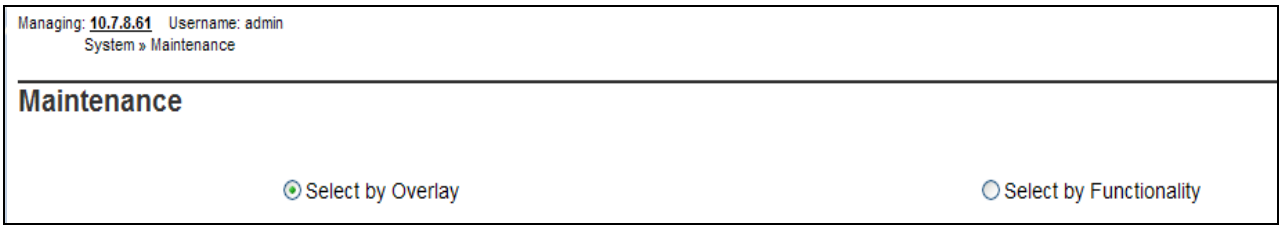


The following screen shows a method to view IP UNIStim telephone status. The screen shows the output of the **Command** "isetShow" in **Group** "Iset". At the time this screen was captured, the first UNIStim telephone listed was involved in an active call with PSTN via the AT&T IP Toll Free service.

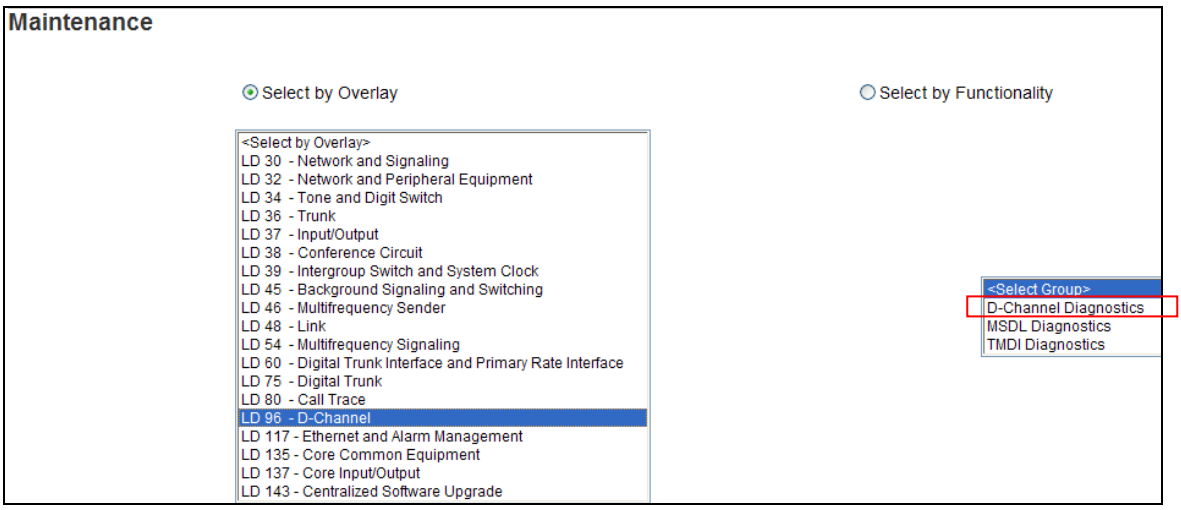


9.2.2. System Maintenance Commands

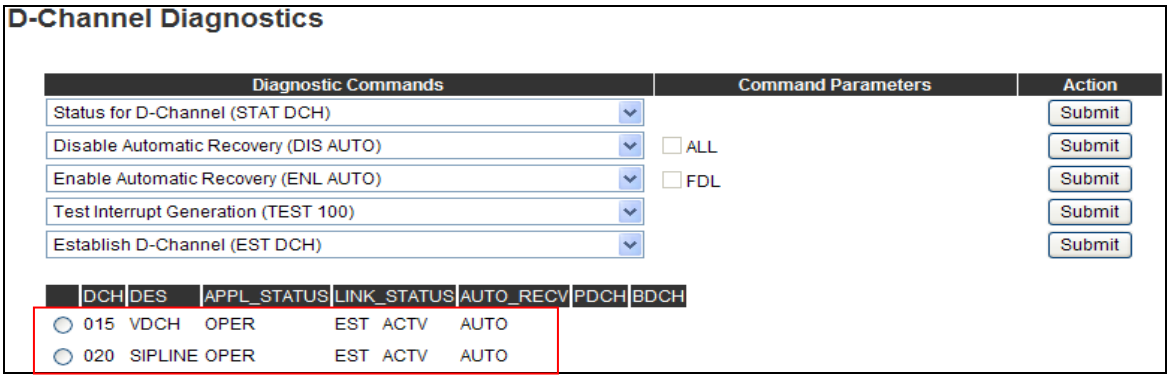
A variety of system maintenance commands are available by navigating to **System → Maintenance** using Element Manager. The user can navigate the maintenance commands using either the "**Select by Overlay**" method or the "**Select by Functionality**" method.



The following screen shows an example where "**Select by Overlay**" has been chosen. The various overlays are listed, and the "**LD 96 – D-Channel**" is selected.



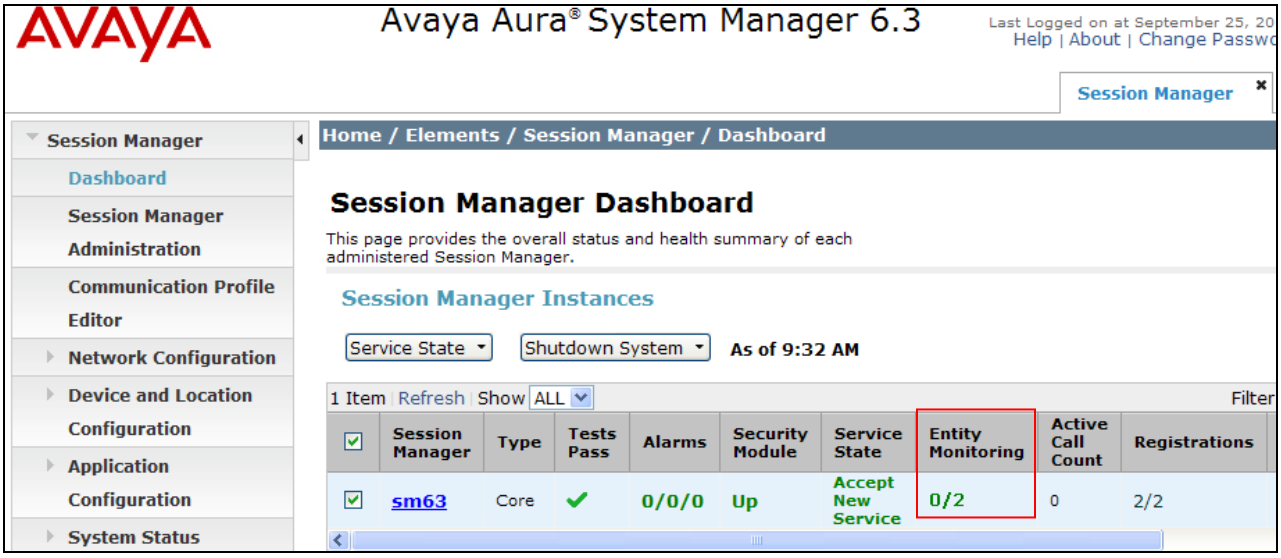
On the preceding screen, if "**LD 96 - D-Channel"** is selected on the left menu with "**D-Channel Diagnostics**" selected on the right menu, a screen such as the following is displayed. D-Channels **15** (SIP GW) and **20** (SIPLINE), show as established (**EST**) and active (**ACTV**).



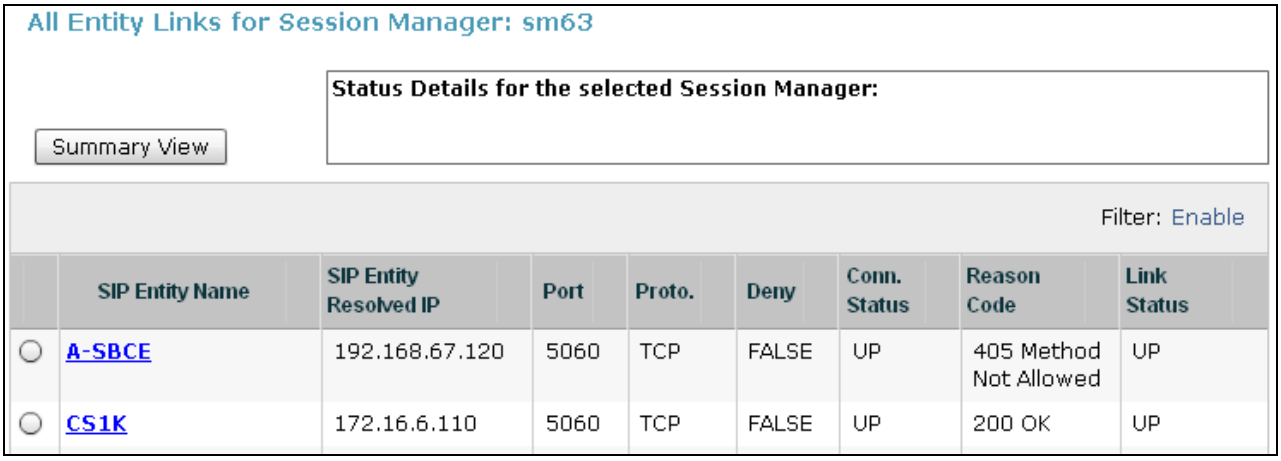
9.3. System Manager / Session Manager Verification

9.3.1. Verify Service State and Entity Link Status

Log in to System Manager. Expand **Elements → Session Manager** and the Dashboard screen is displayed. Verify that the **Service State** column shows "**Accept New Service**", and the **Entity Monitoring** column shows "**0**" Entities are down.



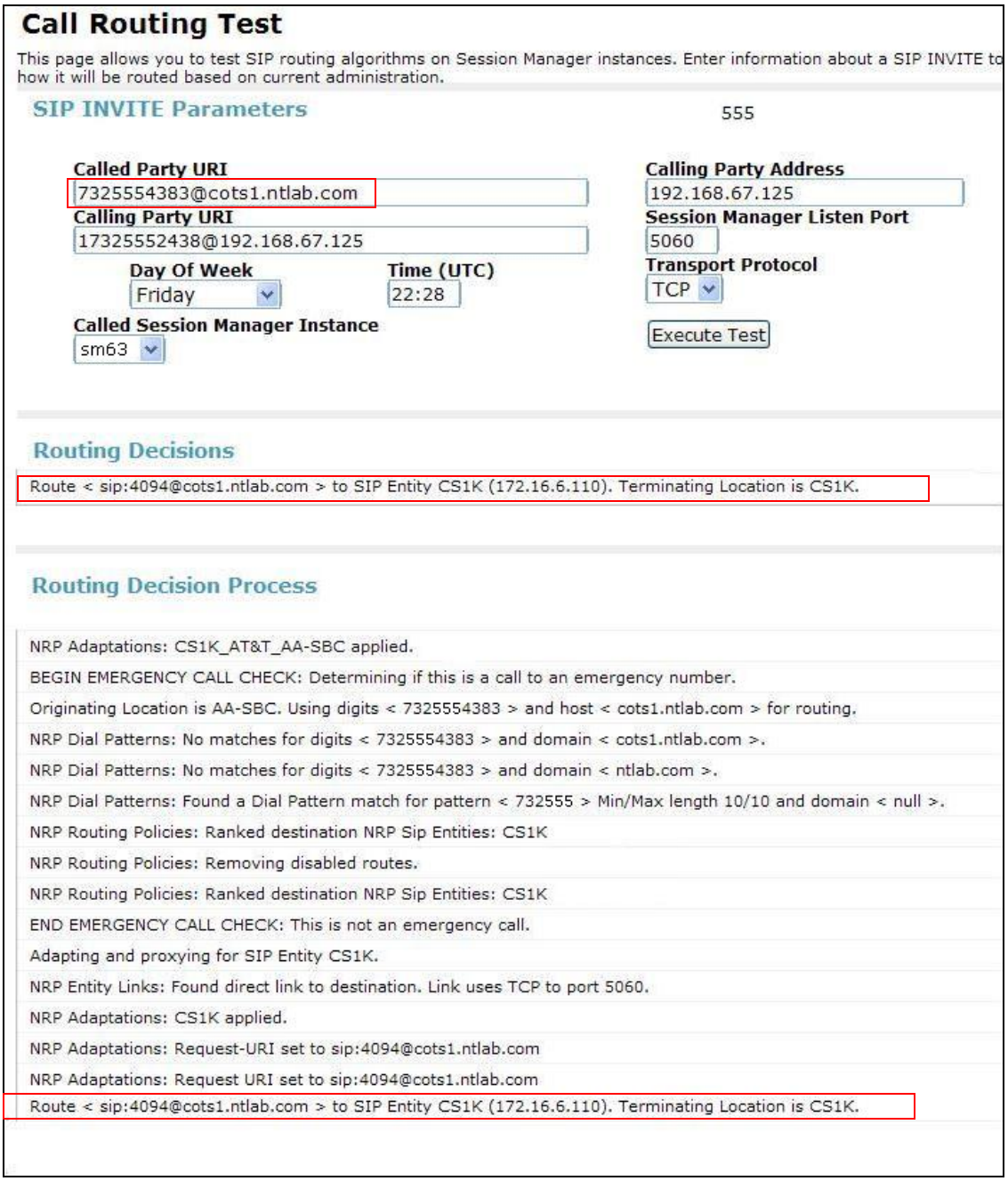
Click on the **Entity Monitoring** display (e.g., **0/2**), and a list of all the provisioned SIP Entities, and their states, are displayed. Under normal operating conditions, the **Conn. Status** should be "**Up**" as shown in the example screen below. The **Reason Code** column indicates that the Avaya SBCE has responded to SIP OPTIONS from Session Manager with a SIP 405 message which is sufficient for SIP Link Monitoring to consider the link up.



9.3.2. Call Routing Test

The Call Routing Test verifies the routing for a particular source and destination. To run the routing test, expand **Elements → Session Manager → System Tools → Call Routing Test**.

The following screen shows an example call routing test for an inbound call to the Avaya CS1000E from PSTN/AT&T.



9.4. Protocol Traces

This section illustrates an example inbound call from PSTN/AT&T IP Toll Free service to DNIS 7325551234 (this number is associated with a CS1000E Directory Number 4096).

1. The following Wireshark trace was captured on the **public** side of the Avaya SBCE (to AT&T), filtered on SIP messages. The INVITE message sent by AT&T to the Avaya SBCE is selected. As can be observed in the example below:
   - The AT&T IP Toll Free service sends the INVITE with the DNIS number **7325551234** in the R-URI; however the Billing number **8885555821** is in the TO header, (used for all calls to this customer).
   - Note that the **maxptime=30** parameter is specified with no ptime parameter.

```
Filter: sip                                   Expression...  Clear  Apply
No.  Time     Source        Destination     Protocol  Info
11   27.229   10.10.10.10   192.168.64.130  SIP/SDP   Request: INVITE sip:7325551234@192.168.64.130:5060, with
⊞ Frame 11: 1082 bytes on wire (8656 bits), 1082 bytes captured (8656 bits)
⊞ Ethernet II, Src: Cisco_29:e4:a0 (a4:93:4c:29:e4:a0), Dst: Intel_31:1b:e9 (90:e2:ba:31:1b:e9)
⊞ Internet Protocol Version 4, Src: 135.25.29.74 (135.25.29.74), Dst: 135.16.170.55 (135.16.170.55)
⊞ User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
⊟ Session Initiation Protocol
  ⊞ Request-Line: INVITE sip:7325551234@192.168.64.130:5060 SIP/2.0
  ⊟ Message Header
    ⊞ Via: SIP/2.0/UDP    10.10.10.10:5060;branch=z9hG4bK9rekqi00400hlrgeu070.1
      MIME-Version: 1.0
      Supported: replaces
      Allow: INVITE, BYE, ACK, CANCEL, PRACK, INFO, REFER
      Accept: application/sdp
      Accept: application/isup
      Accept: application/dtmf
      Accept: application/dtmf-relay
      Accept: multipart/mixed
      Privacy: none
    ⊞ P-Asserted-Identity: <sip:7325551000@10.10.10.10  >
    ⊞ From: <sip:8185551111@10.10.10.10  >;isup-oli=00;tag=905771306713283_c1b09.1.1.1379097345250.0_296_1141
    ⊞ To: <sip:8885551111@192.168.64.130>
      Call-ID: 47033942349966296@c1b09_1_1
      Max-Forwards: 66
    ⊞ CSeq: 2 INVITE
      Content-Type: application/sdp
      Content-Length: 309
    ⊞ Contact: <sip:10.10.10.10  :5060;transport=udp>
  ⊟ Message Body
    ⊟ Session Description Protocol
        Session Description Protocol Version (v): 0
      ⊞ Owner/Creator, Session Id (o): Sonus_UAC 13439 1270 IN IP4 10.10.10.10
        Session Name (s): SIP Media Capabilities
      ⊞ Connection Information (c): IN IP4 10.10.10.10
      ⊞ Time Description, active time (t): 0 0
      ⊞ Media Description, name and address (m): audio 17950 RTP/AVP 18 0 2 100
      ⊞ Media Attribute (a): rtpmap:18 G729/8000
      ⊞ Media Attribute (a): fmtp:18 annexb=no
      ⊞ Media Attribute (a): rtpmap:0 PCMU/8000
      ⊞ Media Attribute (a): rtpmap:2 G726-32/8000
      ⊞ Media Attribute (a): rtpmap:100 telephone-event/8000
      ⊞ Media Attribute (a): fmtp:100 0-15
        Media Attribute (a): sendrecv
      ⊞ Media Attribute (a): maxptime:30
```

2. The following trace captured on the **private** side of the Avaya SBCE (to the CPE), filtered on SIP messages. The same INVITE message sent by AT&T is selected, though it is now sent by the Avaya SBCE to Session Manager (then to the CS1000E). As can be observed in the example below:

   - The contents of the R-URI (**7325551234**) have been copied into the TO header, and the **maxptime=30** parameter has been changed to **ptime=30** by the signaling manipulation defined in **Section 8.3.9**.

```
No.  Time        Source          Destination     Protocol  Length  Info
111  11.448091   192.168.67.120  192.168.67.47   SIP/SDP   1131    Request: INVITE sip:0000011051@

⊞ Frame 111: 1131 bytes on wire (9048 bits), 1131 bytes captured (9048 bits)
⊞ Ethernet II, Src: Intel_31:1b:ed (90:e2:ba:31:1b:ed), Dst: Ibm_40:56:90 (e4:1f:13:40:56:90)
⊞ Internet Protocol Version 4, Src: 192.168.67.120 (192.168.67.120), Dst: 192.168.67.47 (192.168.67.47)
⊞ Transmission Control Protocol, Src Port: 13486 (13486), Dst Port: sip (5060), Seq: 2, Ack: 1, Len: 1077
⊟ Session Initiation Protocol
  ⊞ Request-Line: INVITE sip:7325551234@cots1.ntlab.com>IP/2.0
  ⊟ Message Header
    ⊞ From: <sip:8185551111@cots1.ntlab.com>isup-oli=00;tag=905771306713283_c1b09.1.1.1379097345250.0_296_1141
    ⊞ To: <sip:7325551234@cots1.ntlab.com>
    ⊞ CSeq: 2 INVITE
      Call-ID: 49717a26ace85b720d87c06955e0d647
    ⊞ Contact: <sip:192.168.67.120:5060;transport=tcp>
      Record-Route: <sip:192.168.67.120:5060;ipcs-line=10425;lr;transport=tcp>
      Allow: INVITE, BYE, ACK, CANCEL, PRACK, INFO, REFER
      Supported: replaces
      Max-Forwards: 65
    ⊞ Via: SIP/2.0/TCP 192.168.67.120:5060;branch=z9hG4bK-s1632-002139069330-1--s1632-
      Accept: application/sdp, application/isup, application/dtmf, application/dtmf-relay, multipart/mixed
      MIME-Version: 1.0
      Privacy: none
    ⊞ P-Asserted-Identity: <sip:7325551000@cots1.ntlab.com>
      Content-Type: application/sdp
      Content-Length: 291
  ⊟ Message Body
    ⊟ Session Description Protocol
        Session Description Protocol Version (v): 0
      ⊞ Owner/Creator, Session Id (o): Sonus_UAC 13439 1270 IN IP4 192.168.67.120
        Session Name (s): SIP
      ⊞ Connection Information (c): IN IP4 192.168.67.120
      ⊞ Time Description, active time (t): 0 0
      ⊞ Media Description, name and address (m): audio 16988 RTP/AVP 18 0 2 100
      ⊞ Media Attribute (a): rtpmap:18 G729/8000
      ⊞ Media Attribute (a): fmtp:18 annexb=no
      ⊞ Media Attribute (a): rtpmap:0 PCMU/8000
      ⊞ Media Attribute (a): rtpmap:100 telephone-event/8000
      ⊞ Media Attribute (a): fmtp:100 0-15
        Media Attribute (a): sendrecv
      ⊞ Media Attribute (a): ptime:30
```

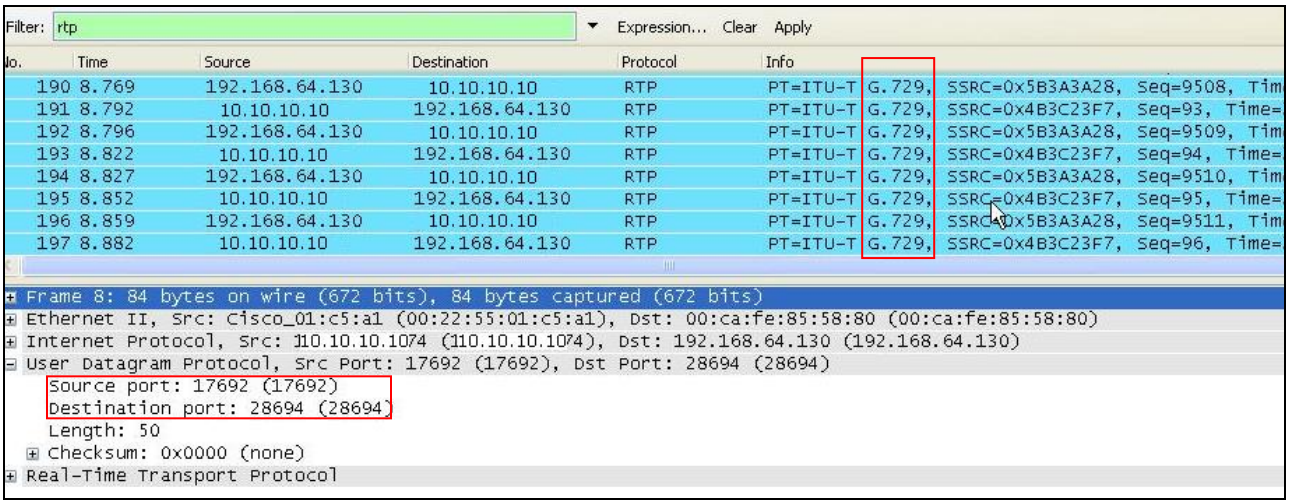
3. The following trace captured on the **private** side of the Avaya SBCE, shows the CS1000E 200 OK response being sent by Session Manager to the Avaya SBCE. As can be observed in the example below:
   - The CS1000E IDT table has an entry for the DNIS number **7325551234** (which converted it to its corresponding CS1000E extension 4096), therefore the CS1000E sends the DNIS number in the P-Asserted-Identity header and the Contact header, (see **Section 2.2.1, Item 5**).
   - Session Manager has added **P-Location**, **AV-Global-Session-ID**, and **P-AV-Message-ID-ID** headers.
   - The CS1000E is sending RFC2833 Telephone event types **100** and **111**.
   - Note that the CS1000E is responding with **ptime:30**.

```
No.  Time        Source          Destination     Protocol  Length  Info
156  13.629308   192.168.67.47   192.168.67.120  SIP/SDP   322     Status: 200 OK, with session description

⊞ Frame 156: 322 bytes on wire (2576 bits), 322 bytes captured (2576 bits)
⊞ Ethernet II, Src: Ibm_40:56:90 (e4:1f:13:40:56:90), Dst: Intel_31:1b:ed (90:e2:ba:31:1b:ed)
⊞ Internet Protocol Version 4, Src: 192.168.67.47 (192.168.67.47), Dst: 192.168.67.120 (192.168.67.120)
⊞ Transmission Control Protocol, Src Port: sip (5060), Dst Port: 13486 (13486), Seq: 3517, Ack: 1079, Len: 268
⊞ [2 Reassembled TCP Segments (1728 bytes): #155(1460), #156(268)]
⊟ Session Initiation Protocol
  ⊞ Status-Line: SIP/2.0 200 OK
  ⊟ Message Header
    ⊞ P-Location: SM;origlocname="Main";origsiglocname="Main";origmedialocname="Main";termlocname="CS1K";termsiglocname="CS1K
    ⊞ P-AV-Message-Id: 1_2
      Server: AVAYA-SM-6.3.5.0.635005
    ⊞ Av-Global-Session-ID: fca550b0-6bfc-11e3-acaf-e41f13326f60
      Allow: INVITE, ACK, BYE, REGISTER, REFER, NOTIFY, CANCEL, PRACK, OPTIONS, INFO, SUBSCRIBE, UPDATE
    ⊞ Contact: <sip:7325551234;phone-context=UnknownUnknown@cots1.ntlab.com:5060;maddr=172.16.6.110;transport=tcp;user=phone;
      Record-Route: <sip:rw-5aabea25@192.168.67.47;lr;transport=TCP>
      Record-Route: <sip:192.168.67.46:15060;transport=tcp;lr;ibmsid=local.1386195439845_843122_843354>
      Record-Route: <sip:rw-5aabea25@192.168.67.47;lr;transport=TCP>
      Record-Route: <sip:192.168.67.120:5060;transport=tcp;lr;ipcs-line=10425>
      Privacy: none
    ⊞ P-Asserted-Identity: <sip:7325551234;phone-context=UnknownUnknown@cots1.ntlab.com;user=phone>
      User-Agent: Nortel CS1000 SIP GW release_7.0 version_ssLinux-7.65.16
      Supported: 100rel, x-nortel-sipvc, replaces
    ⊞ To: <sip:7325551234@customera.com>;tag=3a95078-6e0610ac-13c4-55013-53325-42275032-53325
    ⊞ From: <sip:8185551111@customera.com>;tag=905771306713283_c1b09.1.1.1379097345250.0_296_1141;isup-oli=00
      Call-ID: 49717a26ace85b720d87c06955e0d647
    ⊞ CSeq: 2 INVITE
    ⊞ Via: SIP/2.0/TCP 192.168.67.120:5060;branch=z9hG4bK-s1632-002139069330-1--s1632-
      Content-Type: application/sdp
      Content-Length: 254
  ⊟ Message Body
    ⊟ Session Description Protocol
        Session Description Protocol Version (v): 0
      ⊞ Owner/Creator, Session Id (o): - 2 2 IN IP4 172.16.6.110
        Session Name (s): -
      ⊞ Connection Information (c): IN IP4 172.16.6.115
      ⊞ Time Description, active time (t): 0 0
      ⊞ Media Description, name and address (m): audio 5574 RTP/AVP 18 100 111
      ⊞ Connection Information (c): IN IP4 172.16.6.115
      ⊞ Media Attribute (a): ptime:30
      ⊞ Media Attribute (a): fmtp:18 annexb=no
      ⊞ Media Attribute (a): rtpmap:100 telephone-event/8000
      ⊞ Media Attribute (a): fmtp:100 0-15
      ⊞ Media Attribute (a): rtpmap:111 X-nt-inforeq/8000
        Media Attribute (a): sendrecv
```

4. The following trace captured on the **public** side of the Avaya SBCE, shows the subsequent 200 OK message sent by the Avaya SBCE to AT&T.
   - The Avaya SBCE has removed the following headers (**Section 8.4.3**):
     - **P-Location**, **AV-Global-Session-ID**, and **P-AV-Message-ID**
   - Removed the **Telephone Event Type 111** (**Section 8.3.9**).
   - The Avaya SBCE has set the TO header back to the customers Billing number **8885551111** (**Section 8.3.9**).

```
No.   Time      Source          Destination     Protocol   Info
17    29.413    192.168.64.130  10.10.10.10     SIP/SDP    Status: 200 OK, with session description

⊞ Frame 17: 1224 bytes on wire (9792 bits), 1224 bytes captured (9792 bits)
⊞ Ethernet II, Src: Intel_31:1b:e9 (90:e2:ba:31:1b:e9), Dst: Cisco_29:e4:a0 (a4:93:4c:29:e4:a0)
⊞ Internet Protocol Version 4, Src: 192.168.64.130 (192.168.64.130), Dst: 10.10.10.10    ( 10.10.10.10  )
⊞ User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)
⊟ Session Initiation Protocol
  ⊞ Status-Line: SIP/2.0 200 OK
  ⊟ Message Header
    ⊞ From: <sip:8185551111@10.10.10.10  >;tag=905771306713283_c1b09.1.1.1379097345250.0_296_1141;isup-oli=00
    ⊞ To: <sip:8885551111@192.168.64.130>;tag=3a95078-6e0610ac-13c4-55013-53325-42275032-53325
    ⊞ CSeq: 2 INVITE
      Call-ID: 47033942349966296@c1b09_1_1
    ⊞ Contact: <sip:7325551234;phone-context=UnknownUnknown@192.168.64.130:5060;transport=udp;user=phone;gsid=fca
      Record-Route: <sip:192.168.64.130:5060;ipcs-line=10425;lr;transport=udp>
      Allow: INVITE, ACK, BYE, REGISTER, REFER, NOTIFY, CANCEL, PRACK, OPTIONS, INFO, SUBSCRIBE, UPDATE
      Supported: 100rel, x-nortel-sipvc, replaces
      User-Agent: Nortel CS1000 SIP GW release_7.0 version_ssLinux-7.65.16
    ⊞ Via: SIP/2.0/UDP 10.10.10.10  :5060;branch=z9hG4bK9rekqi00400hlrgeu070.1
      Server: AVAYA-SM-6.3.5.0.635005
      Privacy: none
    ⊞ P-Asserted-Identity: <sip:7325551234;phone-context=UnknownUnknown@192.168.64.130;user=phone>
      Content-Type: application/sdp
      Content-Length: 222
  ⊟ Message Body
    ⊟ Session Description Protocol
        Session Description Protocol Version (v): 0
      ⊞ Owner/Creator, Session Id (o): - 2 2 IN IP4 192.168.64.130
        Session Name (s): -
      ⊞ Connection Information (c): IN IP4 192.168.64.130
      ⊞ Time Description, active time (t): 0 0
      ⊞ Media Description, name and address (m): audio 16900 RTP/AVP 18 100
      ⊞ Connection Information (c): IN IP4 192.168.64.130
      ⊞ Media Attribute (a): ptime:30
      ⊞ Media Attribute (a): fmtp:18 annexb=no
      ⊞ Media Attribute (a): rtpmap:100 telephone-event/8000
      ⊞ Media Attribute (a): fmtp:100 0-15
        Media Attribute (a): sendrecv
```

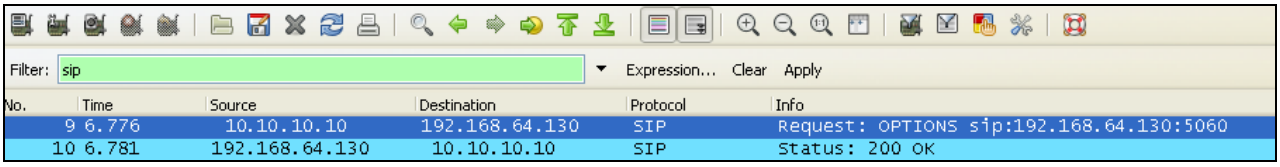
Changing the display filter to **rtp**, the media streams for this call are displayed. Note that the UDP ports used are within the range defined in **Section 8.5.2**. Also note that G.729 was the codec used.



9.5. Avaya Session Border Controller for Enterprise Verification

9.5.1. Verify Avaya SBCE Connectivity to AT&T IP Toll Free

Verify that SIP trunk connection from Avaya SBCE (**192.168.64.130**) to AT&T IP Toll Free border element (**10.10.10.10**[6]) is up and communicating with SIP OPTIONS and response messages. In the example below, AT&T has sent an OPTIONS and Session Manager (via the Avaya SBCE) has responded with 200 OK.



9.5.2. Internal Tracing

Step 1 – Using the left hand column menu described in **Section 8**, navigate to **Device Specific Settings → Troubleshooting → Trace**.

Step 2 - Select the **Packet Capture** tab and select the following:

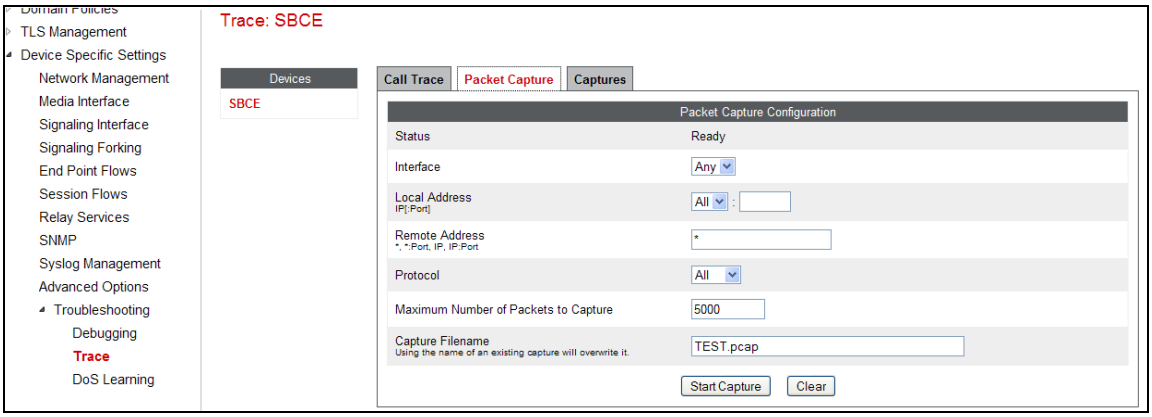
a. Select the desired **Interface** from the drop-down menu. If "**Any**" is selected, then the Avaya SBCE will trace traffic from both the A1 and B1 interfaces.

b. Specify the **Maximum Number of Packets to Capture** (.e.g., **5000**)

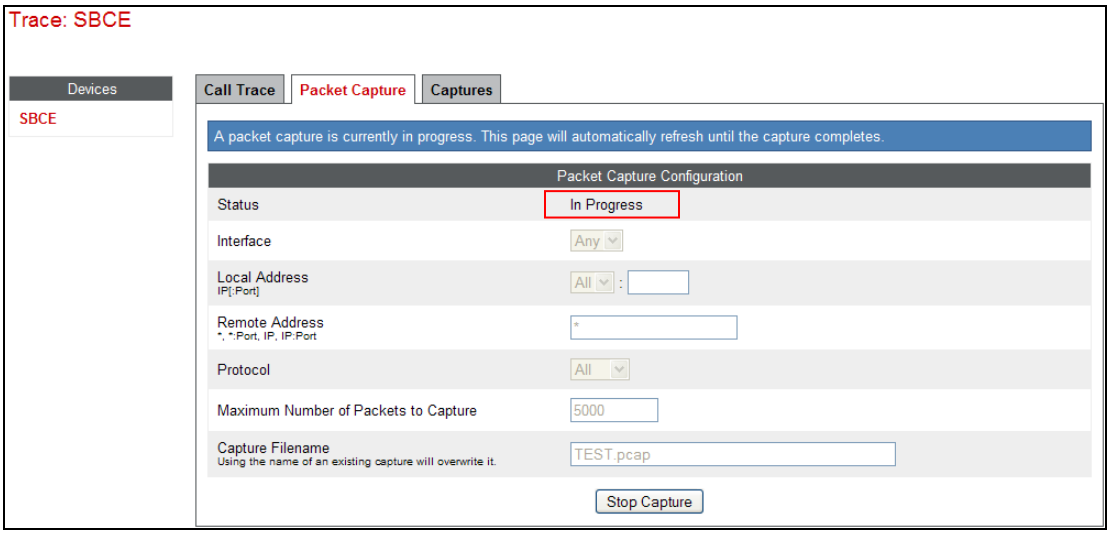
c. Specify a **Capture Filename** (e.g., **TEST.pcap**).

d. Click **Start Capture** to begin the trace.

[6] See the note in **Section 3.1**.



The capture process will initialize, (with the message "Please wait while your settings are saved and the capture is started" displayed), and then the **Status** field will change to "**In Progress**".

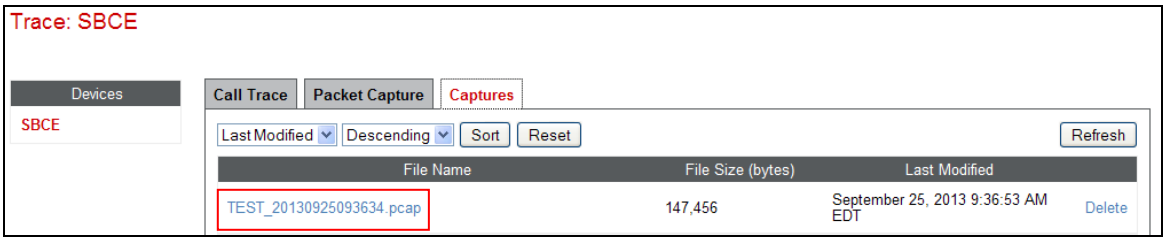


Step 3 – Run the test.

Step 4 – Click on the **Stop Capture** button.

Step 5 - Click on the **Captures** tab and the packet capture is listed as a *.pcap* file with the date and time added to filename.

Step 6 - Click on the file name link to download the file and use an application such as Wireshark to open the trace.



10. Conclusion

As illustrated in these Application Notes, Avaya Aura® Session Manager, Avaya Communication Server 1000E (CS1000E), and Avaya Session Border Controller for Enterprise (Avaya SBCE) can be configured to interoperate successfully with the AT&T IP Toll Free service. This solution provides users of CS1000E the ability to support inbound toll free calls over an AT&T IP Toll Free SIP trunk service connection.

Note: These Application Notes do NOT cover the AT&T IP Transfer Connect service option of the AT&T IP Toll Free service. That solution is *not* supported by the CS1000E.

The reference configuration shown in these Application Notes is representative of a basic enterprise customer configuration and is intended to provide configuration guidance to supplement other Avaya product documentation. It is based upon formal interoperability compliance testing as part of Avaya DevConnect Service Provider program.

11. References

11.1. Avaya

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

Avaya Communication Server 1000E

[1] *Network Routing Service Fundamentals, Avaya Communication Server 1000*, Release 7.6, Document Number NN43001-130, Issue 04.01, March 2013.
[2] *IP Peer Networking Installation and Commissioning, Avaya Communication Server 1000*, Release 7.6, Document Number NN43001-313, Issue 06.01, March 2013.
[3] *Unified Communications Management Common Services Fundamentals, Avaya Communication Server 1000*, Release 7.6, Document Number NN43001-116, Issue 06.01, March 2013.
[4] *SIP Line Fundamentals Avaya Communication Server 1000*, Release 7.6, NN43001-508, Issue 04.01
[5] *Avaya CallPilot® Communication Server 1000 and Avaya CallPilot Server Configuration* 5.1, NN44200-312, 02.01, October 2012

Avaya Aura® Session Manager/System Manager

[6] *Administering Avaya Aura® Session Manager*, Release 6.3, December, 2012
[7] *Implementing Avaya Aura® Session Manager*, Release 6.3, March, 2013
[8] *Implementing Avaya Aura® System Manager*, Release 6.3, Issue 1, December, 2012

Avaya Session Border Controller for Enterprise

[9] *Installing Avaya Session Border Controller for Enterprise*, Release 6.2, Issue 3, June 2013
[10] *Administering Avaya Session Border Controller for Enterprise*, Release 6.2, Issue 2, March 2013

Avaya Aura® Contact Center

[11] *Avaya Aura® Contact Center Server Administration*, NN44400-610, Document issue: 03.02, Document date: 24 August 2011, Product release: Release 6.2
[12] *Avaya Aura® Contact Center Administration–Client Administration*, Release 6.2, NN44400-611, 03.02, 24 August 2011
[13] *Avaya Aura® Contact Center Configuration — Avaya Communication Server 1000 Integration*, NN44400-512, Document issue: 02.03, Document date: 12 November 2010, Product release: Release 6.0/6.1
[14] *Avaya Aura® Contact Center Installation*, Release 6.2, NN44400-311, 03.03, 11 October 2011

[15] *Avaya Aura® Contact Center Commissioning*, Release 6.2, NN44400-312, 03.02, 24 August 2011
[16] *Avaya Aura® Contact Center SIP Commissioning*, NN44400-511, Document issue: 03.02, Document date: 24 August 2011, Product release: Release 6.2

AT&T IP Toll Free Service:

[17] AT&T IP Toll Free Service description - http://www.business.att.com/enterprise/Service/voice-services/contact-center-solutions/ip-toll-free/

12. Addendum 1 – Avaya Session Border Controller for Enterprise Redundancy to Multiple AT&T Border Elements

The AT&T IP Toll Free service may provide multiple network border elements for redundancy purposes. The Avaya SBCE can be provisioned to support this redundancy configuration.

Given two AT&T border elements **10.10.10.10**[7] and **10.10.10.11**, the Avaya SBCE is provisioned as follows to include the backup trunk connection to 10.10.10.11 (the primary trunk connection to 10.10.10.10 is defined in **Sections 8.3.4** and **8.3.6**).

12.1.1. Configure the Secondary Border Element Server Configuration

Repeat the steps in **Section 8.3.6** to create a Server Configuration for the connection to the AT&T secondary Border Element, using the following entries:

Step 1 - In the **Profile Name** window enter a Profile Name (e.g., "**ATT_Secondary_SC**") and select **Next**.
Step 2 – In the **Add Server Configuration Profile - General** window for **Server Type:** select **Trunk Server.**

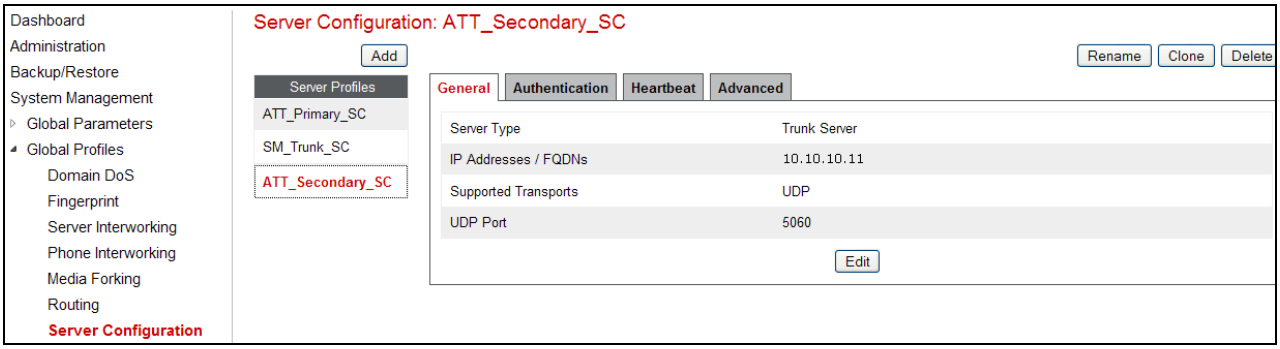
- Enter **IP Address: 10.10.10.11**.
- For **Supported Transports**: check **UDP**
- For **UDP Port:** enter **5060**
- Select **Next**

Step 3 - Accept default values for the **Add Server Configuration Profile - Authentication** and **Heartbeat** windows (not shown).
Step 4 – The **Add Server Configuration Profile - Advanced** window will open.

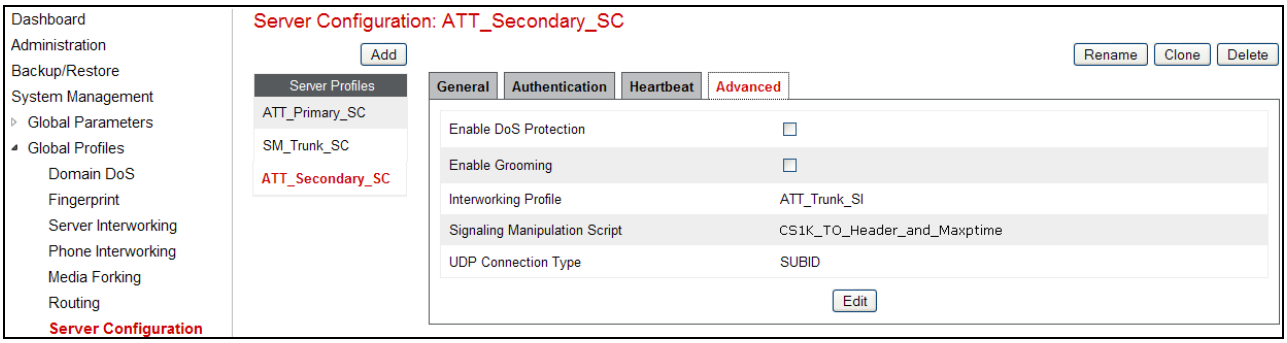
- Select **ATT_SI** for **Interworking Profile** (created in **Section 8.3.2**).
- For the **Signaling Manipulation Script** select the **CS1K_TO_Header_and_Maxptime** script that was defined in **Section 8.3.9**.

Step 5 - Select **Finish**.

The following screen shots show the completed **General** and **Advanced** tabs.



[7] See the note in **Section 3.1**.



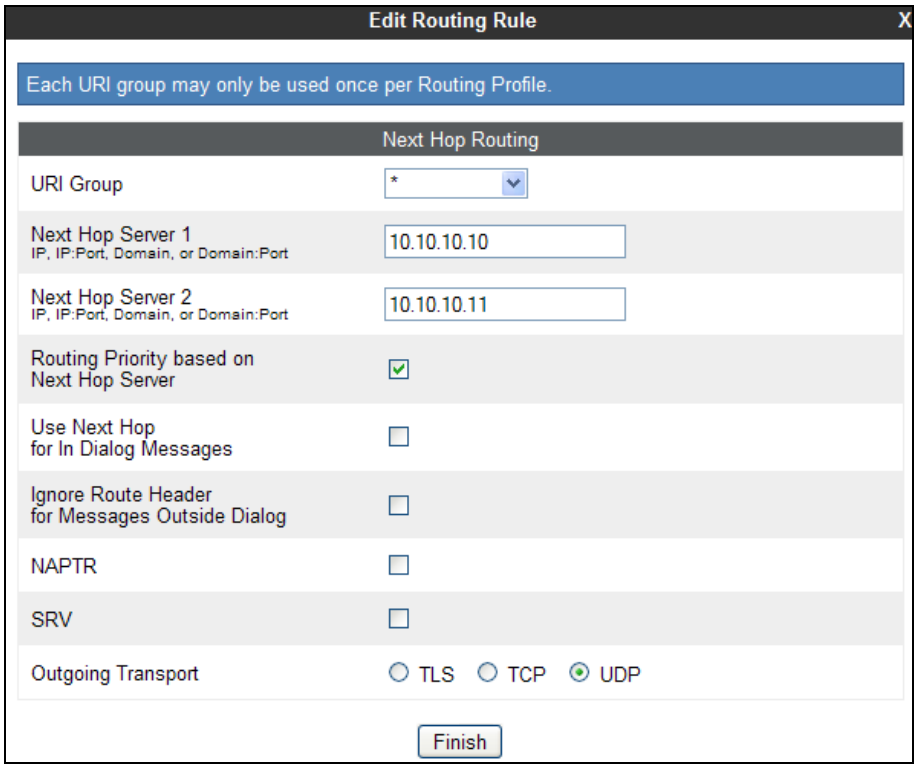
12.1.2. Add Secondary Border Element IP Address to Routing

Repeat the steps in **Section 8.3.4** to add a Routing Profile for the AT&T secondary Border Element.

Step 1 – Select the profile created in **Section 8.3.4** (e.g., **To_ATT_RP)**.
Step 2 - Click **Next**, then enter the following:

- Set **Next Hop Server 2:** to **10.10.10.11**.

Step 3 - Click **Finish**.

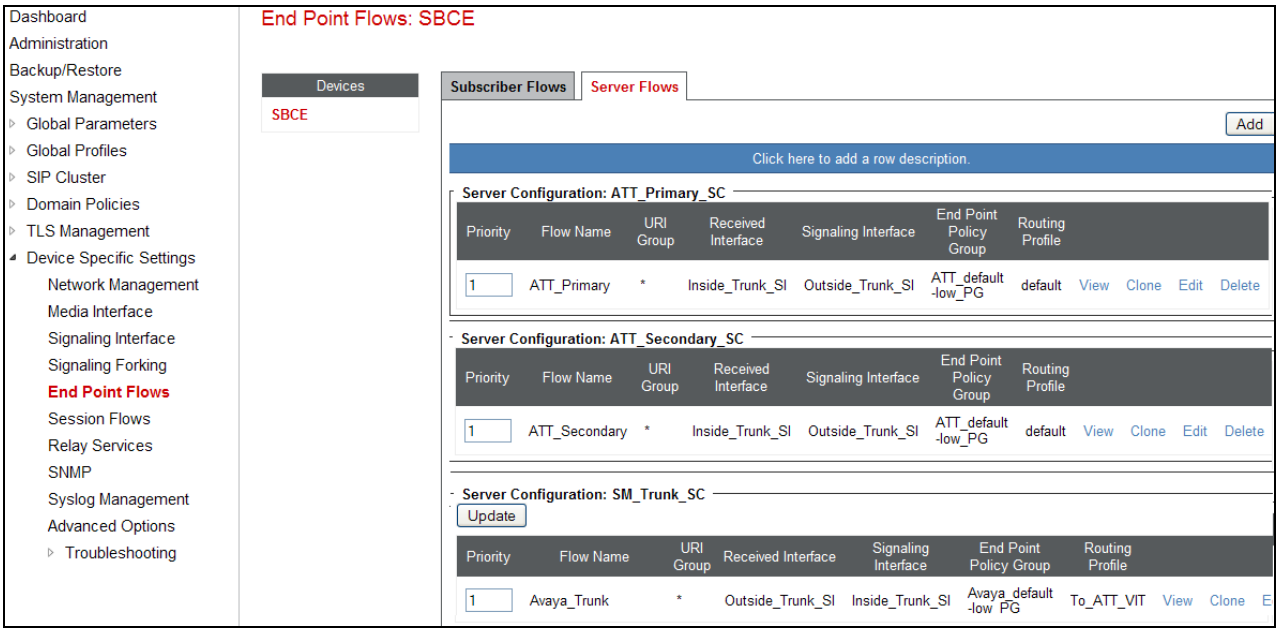


12.1.3. Configure Secondary AT&T Border Element End Point Flow

Step 1 – Repeat the steps in **Section 8.5.5**, with the following changes, to add an Endpoint Flow for the AT&T secondary Border Element:

- For **Name:** enter "**ATT_Secondary**"

Step 4 - Click **Finish** (not shown)



When completed the Avaya SBCE will issue OPTIONS messages to the primary (**10.10.10.10** and secondary (**10.10.10.11**) border elements.

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.