Avaya Solution & Interoperability Test Lab

# Application Notes for Avaya Communication Server 1000E Release 7.5, Avaya Aura® Session Manager 6.1, and Avaya Aura® Session Border Controller 6.0.2 with AT&T IP Flexible Reach SIP Trunk Service – Issue 1.1

## Abstract


These Application Notes illustrate a sample configuration using Avaya Communication Server 1000E Release 7.5, Avaya Aura® Session Manager Release 6.1, and the Avaya Aura® Session Border Controller Release 6.0.2, with the AT&T IP Flexible Reach SIP Trunk service using either **AVPN** or **MIS/PNT** transport connections.

Avaya Aura® Session Manager 6.1 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Communication Server 1000E 7.5 is a telephony server, and is the point of connection between the enterprise endpoints and Avaya Aura® Session Manager. An Avaya Aura® Session Border Controller 6.0.2 is the point of connection between Avaya Aura® Session Manager and the AT&T IP Flexible Reach service, and is used to not only secure the SIP trunk, but also to make adjustments to the signaling for interoperability.

The AT&T IP Flexible Reach service is one of several SIP-based Voice over IP (VoIP) services offered to enterprises for a variety of voice communications needs. The AT&T IP Flexible Reach service allows enterprises in the U.S.A. to place outbound local and long distance calls, receive inbound Direct Inward Dialing (DID) calls from the PSTN, and place calls between an enterprise's sites.

AT&T is a member of the 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.

# Table of Contents

# 1. Introduction

These Application Notes illustrate a sample configuration using Avaya Communication Server 1000E Release 7.5 (CS1000E), Avaya Aura® Session Manager Release 6.1 (Session Manager), and the Avaya Aura® Session Border Controller Release 6.0 (Avaya Aura® SBC), with the AT&T IP Flexible Reach SIP trunk service for PSTN access.

The AT&T IP Flexible Reach service is one of several SIP-based Voice over IP (VoIP) services offered to enterprises for a variety of voice communications needs. The AT&T IP Flexible Reach service allows enterprises in the U.S.A. to place outbound local and long distance calls, receive inbound Direct Inward Dialing (DID) calls from the PSTN, and place calls between an enterprise's sites. The AT&T IP Flexible Reach service utilizes AVPN[1] or MIS/PNT[2] transport services.

For more information on the, AT&T IP Flexible Reach service visit: http://www.business.att.com/enterprise/Service/business-voip-enterprise/network-based-voip-enterprise/ip-flexible-reach-enterprise/.

# 2. General Test Approach and Test Results

The interoperability compliance testing focused on verifying inbound and outbound call flows (see **Section 2.3** for examples) between the CS1000E, the Avaya Aura® SBC, and the AT&T IP Flexible Reach service. The CS1000E users make calls to and from the PSTN via the AT&T IP Flexible Reach service.

## 2.1. Interoperability Compliance Testing

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

- SIP trunking of inbound and outbound calls.
  - Incoming calls from the PSTN were routed to the DID numbers assigned by the AT&T IP Flexible Reach service to the CS1000E location. These incoming PSTN calls arrived via the SIP Trunk and were answered by Avaya IP UNIStim telephones and fax machine emulation software (Ventafax). Proper call disconnect was verified.
  - Outgoing calls from the CS1000E location to the PSTN were routed via the SIP Trunk to the AT&T IP Flexible Reach service. These outgoing PSTN calls were originated from Avaya IP UNIStim telephones, and fax machine emulation software (Ventafax). Proper call disconnect was verified.
  - Use of G.729A and G.711Mu codecs were verified.
- Inbound and outbound T.38 Fax, using combinations of G3 and SG3 modes, were verified.

[1] AVPN uses compressed RTP (cRTP).
[2].MIS/PNT does not support cRTP.

- CS1000E station call coverage to Avaya Call Pilot® for message generation and retrieval (including Message Wait Indicator).
- Passing of DTMF events (RFC2833) and their recognition by navigating automated menus (e.g. Avaya Call Pilot® message selection and retrieval)
- PBX features such as hold, resume, conference and transfer.
- Requests for privacy (i.e., caller anonymity) for CS1000E outbound calls to the PSTN, and for inbound calls from the PSTN to CS1000E, were verified.
- SIP OPTIONS monitoring of the health of the SIP trunk was verified. Both the AT&T IP Flexible Reach service and the Avaya Aura® SBC were able to monitor health using SIP OPTIONS.
- Inbound calls to CS1000E station that were call forwarded back to PSTN destinations, through use of Diversion Header were verified.
- Proper UDP port ranges for RTP media (16384-32767) were verified.

## 2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results. The following observations were noted during testing:

### 2.2.1 Known Limitations

1. To allow the CS1000E user to transfer a call from PSTN user A to PSTN user B, before user B has answered the call (unattended transfer), CS1000E plug-in 501 must be enabled as shown in **Section 5.7**. While plug-in 501 will allow the CS1000E user to complete the transfer operation, user A will not hear ring back tone while user B is ringing in this case. PSTN users A and B will have two-way talk path once user B answers.

2. G.711 fax is not supported in the reference configuration. T.38 faxing is supported, as is Group 3 and Super Group 3 fax. Fax speeds to 14400 bps are supported in the configuration tested. In addition, Fax Error Correction Mode (ECM) is supported in the reference configuration.

3. The AT&T IP Flexible Reach service does not support SIP History-Info headers. However, the AT&T IP Flexible Reach service requires that SIP Diversion Header be sent for certain redirected calls (e.g. Call Forward). Session Manager can convert the History Info header into the Diversion Header by the use of the adaptation "*DiversionTypeAdapter*" for these types of calls (see **Section 6.3.2)**. For all other calls, the Avaya Aura® SBC will strip off History-Info headers (see **Section 7.2.5**).

4. Emergency 911/E911 Services Limitations and Restrictions – Although AT&T provides 911/E911 calling capabilities, AT&T does not warrant or represent that the equipment and software (e.g., IP PBX) reviewed in this customer configuration guide will properly operate with AT&T IP Flexible Reach to complete 911/E911 calls; therefore, it is the customer's responsibility to ensure proper operation with the equipment/software vendor.

   While AT&T IP Flexible Reach services support E911/911 calling capabilities under certain Calling Plans, there are circumstances when the E911/911 service may not be available, as stated in the Service Guide for AT&T IP Flexible Reach found at http://new.serviceguide.att.com. Such circumstances include, but are not limited to, relocation of the end user's CPE, use of a non-native or virtual telephone number, failure in the broadband connection, loss of electrical power, and delays that may occur in updating the Customer's location in the automatic location information database. Please review the AT&T IP Flexible Reach Service Guide in detail to understand the limitations and restrictions.

## 2.3. Call Flows

To understand how inbound AT&T IP Flexible Reach service calls are handled by the Avaya CPE environment, three basic call flows are described in this section. However, for brevity, not all possible call flows are described.

### 2.3.1 Inbound

The first call scenario illustrated is an inbound AT&T IP Flexible Reach service call that arrives at the Avaya Aura® SBC, to Session Manager, and is subsequently routed to the CS1000E, which in turn routes the call to a phone or fax.

1. A PSTN phone originates a call to an AT&T IP Flexible Reach service number.
2. The PSTN routes the call to the AT&T IP Flexible Reach service network.
3. The AT&T IP Flexible Reach service routes the call to the Avaya Aura® SBC.
4. The Avaya Aura® SBC performs SIP Network Address Translation (NAT) and any necessary SIP header modifications, and routes the call to Session Manager.
5. Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Routing Policies, determines to 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 a phone or fax.

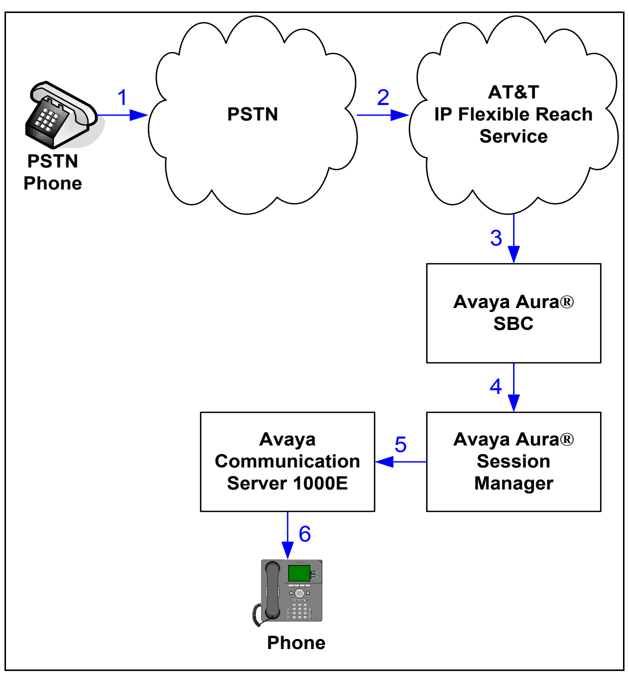



**Figure 1 - Inbound AT&T IP Flexible Reach Call**

## 2.3.2 Outbound

The second call scenario illustrated is an outbound call initiated on CS1000E, routed to Session Manager and is subsequently sent to the Avaya Aura® SBC for delivery to AT&T IP Flexible Reach service.

1. A CS1000E phone or fax originates a call to an AT&T IP Flexible Reach service number for delivery to PSTN.
2. CS1000E routes the call to the Session Manager.
3. Session Manager applies any necessary SIP header adaptations and digit conversions, and based on configured Routing Policies, determines to where the call should be routed next. In this case, Session Manager routes the call to the Avaya Aura® SBC.
4. The Avaya Aura® SBC performs SIP Network Address Translation (NAT) and any necessary SIP header modifications, and routes the call to the AT&T IP Flexible Reach service.
5. The AT&T IP Flexible Reach service delivers the call to PSTN.

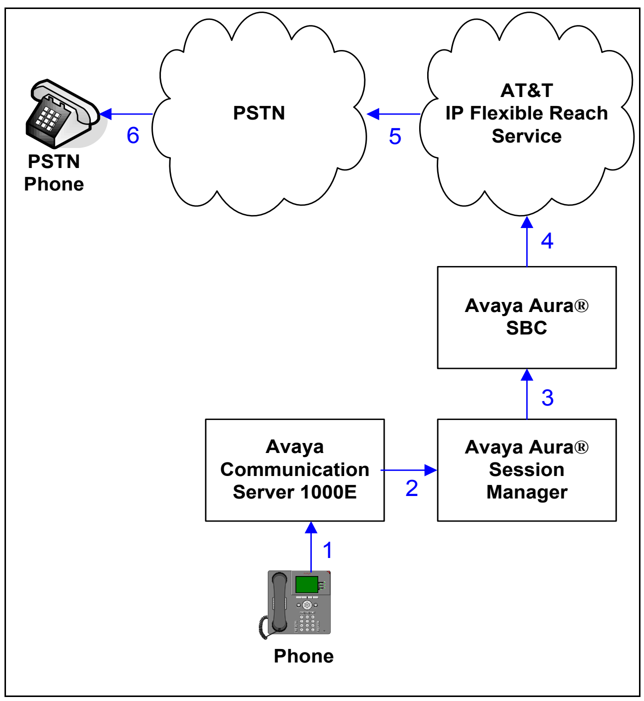



**Figure 2 - Outbound AT&T IP Flexible Reach Call**

## 2.3.3 Call Forward Re-direction

The third call scenario illustrated is an inbound AT&T IP Flexible Reach service call that arrives at the Avaya Aura® SBC, to Session Manager, and subsequently CS1000E. The CS1000E routes the call to a destination station, however the station has set Call Forwarding to an alternate destination. Without answering the call, CS1000E immediately redirects the call back to the AT&T IP Flexible Reach service for routing to the alternate destination.

> **Note** – In cases where calls are forwarded to an alternate destination such as an N11, NPA-555-1212, or 8xx numbers, then the AT&T IP Flexible Reach service requires the use of SIP Diversion Header for the redirected call to complete (see **Section 6.3.2**).

1. Same as the first call scenario in **Section 2.3.1**.
2. Because the CS1000E phone has set Call Forward to another AT&T IP Flexible Reach service number, CS1000E initiates a new call back out to Session Manager, the Avaya Aura® SBC, and to the AT&T IP Flexible Reach service network.
3. The AT&T IP Flexible Reach service places a call to the alternate destination and upon answering; CS1000E connects the calling party to the target party.

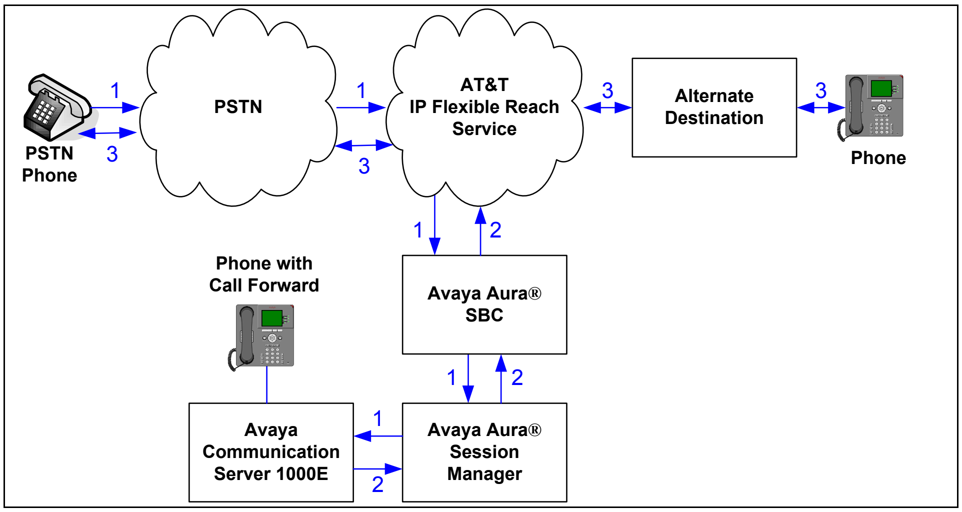



**Figure 3 - Re-directed (e.g. Call Forward) AT&T IP Flexible Reach Call**

## 2.3.4 Coverage to Voicemail

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

1. Same as the first call scenario in **Section 2.3.1**.
2. The called CS1000E 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.

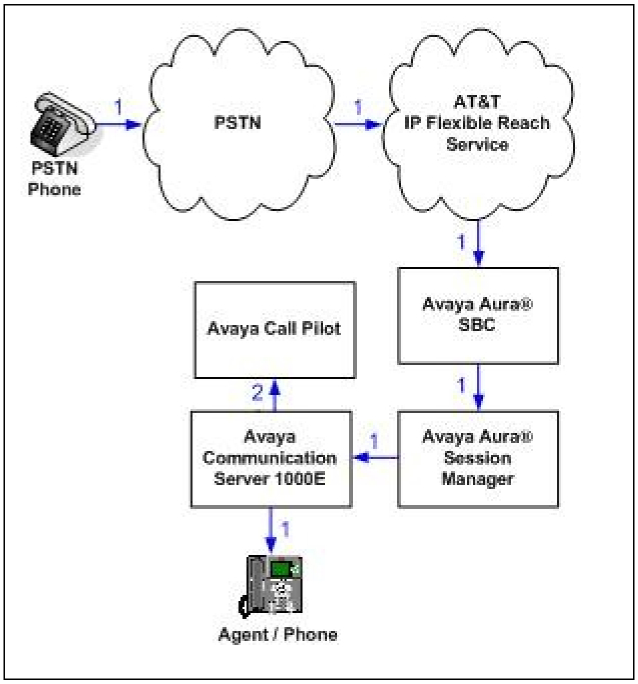



**Figure 4 - Coverage to Voicemail**

## 2.4. Support

### 2.4.1 Avaya

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

### 2.4.2 AT&T

AT&T customers may obtain support for the AT&T IP Flexible Reach service by calling (877) 288-8362.

# 3. Reference Configuration

The reference configuration used in these Application Notes is shown in **Figure 5** and consists of several components:

- 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 Consumer Off The Shelf (COTS) server
    - SIP Signaling Server

   - Avaya Unified Communications Management (UCM)

  **Note** – Only the CS1000E system provisioning providing SIP trunk functionality is described in these application notes. For additional CS1000E system provisioning documentation see **Section 12.**

- Avaya "desk" phones are represented with Avaya 1140E and 2004 UNIStim IP phones.

- The Avaya Aura® SBC provides address translation and SIP header manipulation between the AT&T IP Flexible Reach service and the enterprise internal network. TCP transport protocol is used between the Avaya Aura® SBC and Session Manager. UDP transport protocol is used between the Avaya Aura® SBC and the AT&T IP Flexible Reach 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 [11] for more information).

- Outbound calls were originated from a phone or fax provisioned on the CS1000E system. SIP signaling is passed from CS1000E system to Session Manager, and to the Avaya Aura® SBC, before being sent to the AT&T network for termination. Media was sent from the calling IP phone directly to the Avaya Aura® SBC. Legacy devices such as analog fax send their audio from the MGC to the Avaya Aura® SBC. The Avaya Aura® SBC then directs the media to the AT&T network.

- Inbound calls were sent from PSTN/AT&T, through the Avaya Aura® SBC to Avaya Aura® Session Manager, and on to the CS1000E system. The CS1000E system terminates the calls to the appropriate phone or fax extensions.

**Note** – In the reference configuration TCP (port 5060) is used as the transport protocol between the CS1000K, the Avaya Aura® SBC, and Session Manager. This was done to facilitate protocol trace analysis. However, Avaya best practices call for TLS (port 5061) to be used as the transport protocol where applicable.

UDP transport using port 5060 is required by the AT&T IP Flexible Reach service for the connection between the Avaya Aura® SBC and the AT&T T IP Flexible Reach border element.

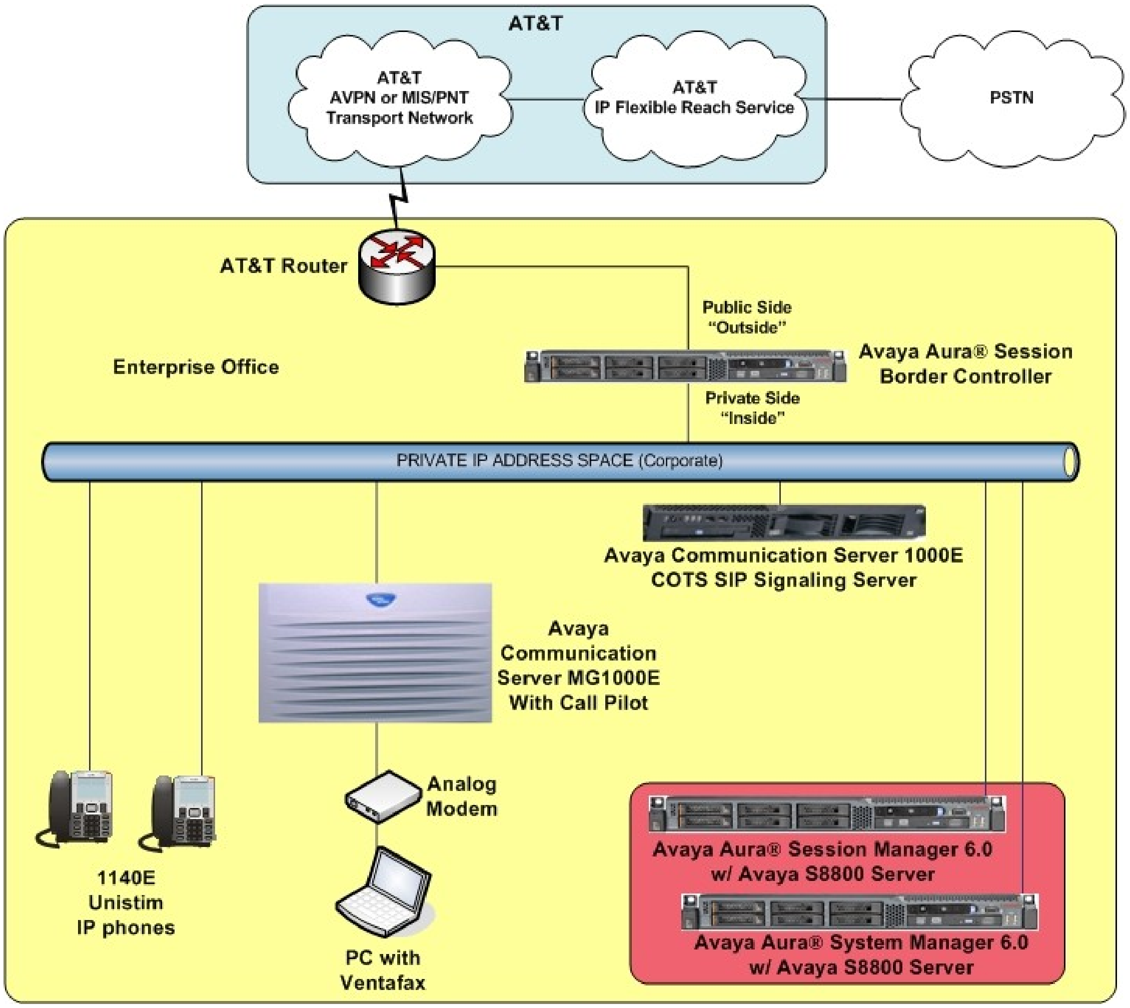



**Figure 5: Avaya Interoperability Test Lab Configuration**

## 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 specific configurations.

**Note** - The AT&T IP Flexible Reach service border element IP address shown in this document is an example. AT&T Customer Care will provide the actual IP addressing as part of the IP Flexible Reach provisioning process.

| Component | Illustrative Value in these Application Notes |
|---|---|
| **Avaya CS1000E** | |
| SIP Signaling Server IP Address (TLAN) | 172.16.6.110 |
| MGC Media (DSP) IP Address (TLAN) | 172.16.6.115 |
| CS1000E extensions | 40xx |
| **Avaya Call Pilot®** | |
| Call Pilot Application | 192.168.67.130 |
| Call Pilot Mailboxes | 4xxx |
| **Acme SBC** | |
| IP Address of "Outside" (Public) Interface (connected to AT&T Access Router/IP Flexible Reach Service) | 192.168.64.130 |
| IP Address of "Inside" (Private) Interface (connected to Session Manager) | 192.168.67.125 |
| **AT&T IP Flexible Reach Service** | |
| Border Element IP Address | 135.25.29.74 |
| AT&T Access router interface (to Acme outside) | 192.168.64.254 |
| AT&T Access Router NAT address (Acme outside address) | 135.16.170.55 |

**Table 1: Illustrative Values Used in these Application Notes**

# 4. Equipment and Software Validated

The following equipment and software were used in the sample configuration.

| Equipment | Software |
|---|---|
| Avaya Communication Server 1000E Platform:<br>• MG1000E Media Gateway<br>• IBM xSeries 306M (COTS) SIP Signaling server | Release 7.5, Version 7.50.17 with Service_Pack_Linux_7.50_17_20110426 and Plug-in 501 Enabled |
| Avaya Call Pilot® | CP 5.00.41 CP50041SU09S, CP500S09G25S, CP500S09G32C |
| Avaya S8800 Server (System Manager) | Avaya Aura® System Manager Release 6.1.0 with SP2 (Build Number 6.1.0.0.7345-6.1.5.106) |
| Avaya S8800 Server (Session Manager) | Avaya Aura® Session Manager Release 6.1 SP2 (Load 6.1.2.0.612004) |
| Avaya S8800 Server (Session Border Controller) | Avaya Aura® Session Border Controller Release 6.0.2.0.3 |
| Avaya 1140E Series IP Deskphones (UNIStim) | FW 0625C8A |
| Avaya 2004 Series IP Deskphones (UNIStim) | FW 0604DCN |
| Fax device | Ventafax Home Version 6.3.102.288 |
| AT&T IP Flexible Reach Service via AVPN or MIS/PNT transport service connections. | VNI 21 |

**Table 2: Equipment and Software Used in the Sample Configuration**

# 5. Configure Avaya Communication Server 1000E

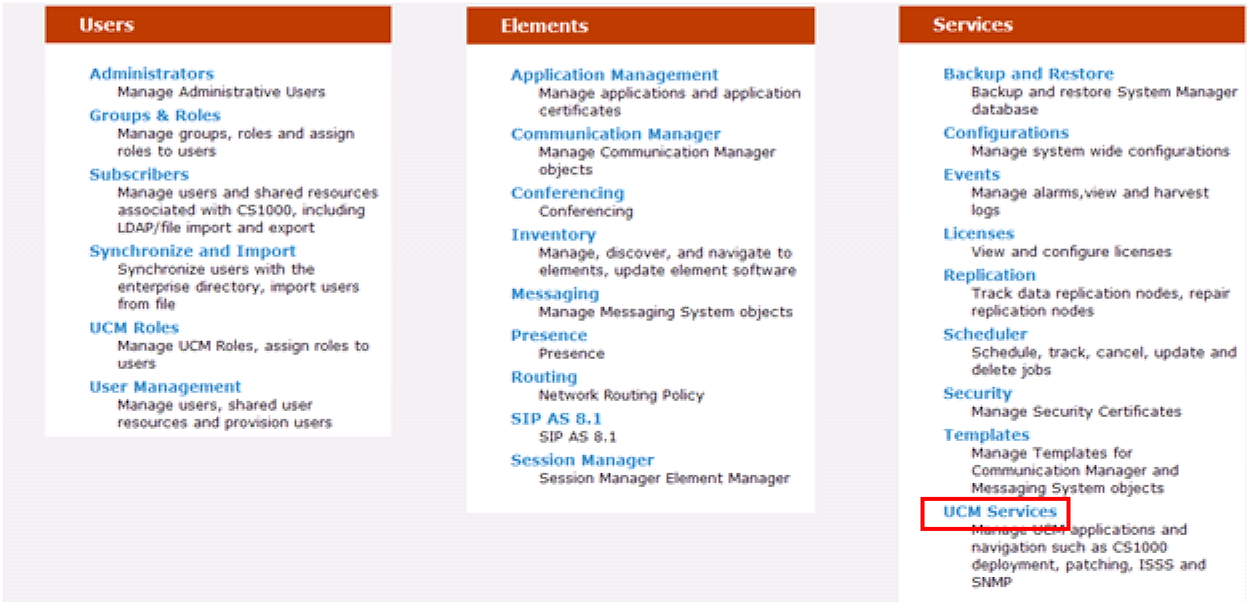
This section describes the Avaya Communication Server 1000E configuration, focusing on the routing of calls to Session Manager over a SIP trunk. In the sample configuration, Avaya Communication Server 1000E Release 7.5 was deployed with Call Server applications running on a CP+PM server platform with MGC, and utilizing a separate SIP Signaling Server.

Avaya Aura® Session Manager Release 6.1 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 Avaya Communication Server 1000E and Session Manager Release 6.1. Therefore NRS was not included in the reference configuration.

This section focuses on the SIP Trunking configuration. 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 Avaya Communication Server 1000E is configured to support analog and UNIStim telephones (although supported, SIP telephones were not part of the reference configuration. For references on how to administer these functions of Avaya Communication Server 1000E, see **Section 12.**

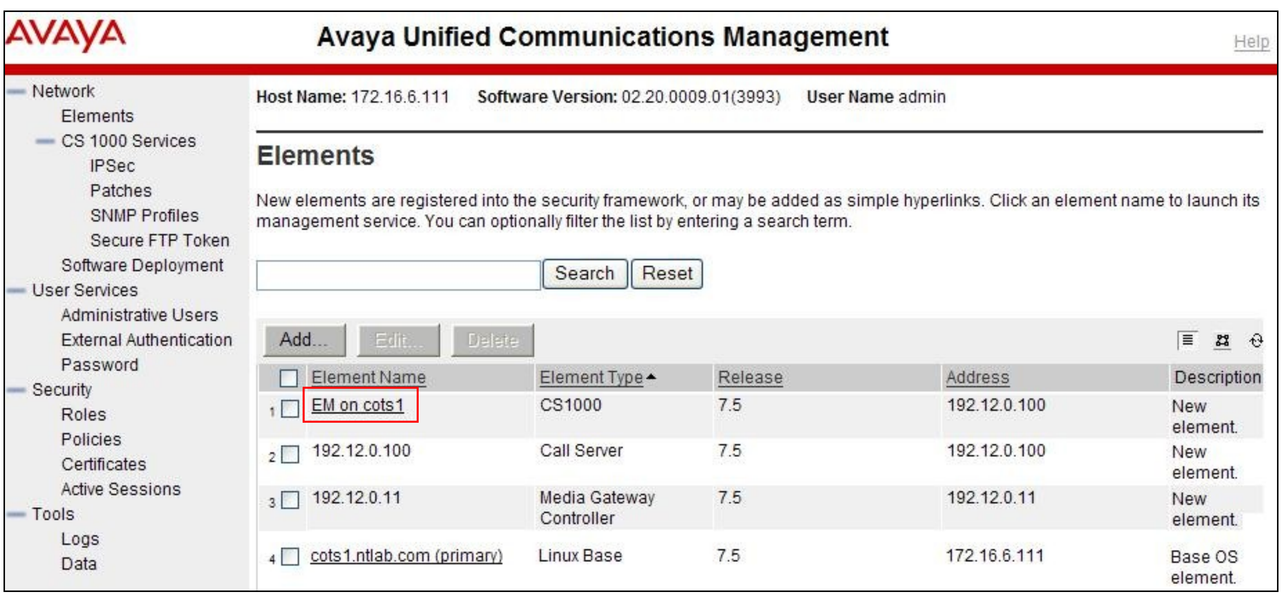
**Step 1** - Unless otherwise noted, all CS1000E provisioning was performed via the Avaya Unified Communication Management (AUCM) web interface. The **AUCM** web interface may be launched directly via **https://<ip address>** where the relevant <ipaddress> in the sample configuration is 172.16.6.111.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


**Note** – Although not used in the reference configuration, Avaya Aura® System Manager may be configured as the Primary Security Server for the Avaya Unified Communications Management application and Avaya Communication Server 1000E is registered as a member of the System Manager Security framework. The Element Manager then may be accessed via System Manager. In this case, access the web based GUI of Avaya Aura® System Manager by using the URL **"http://<ip-address>/SMGR"**, where **<ip-address>** is the IP address of Avaya Aura® System Manager. Log in with appropriate credentials. The Avaya Aura® System Manager Home Page will be displayed. Under the **Services** category on the right side of the page, click the **UCM Services** link.



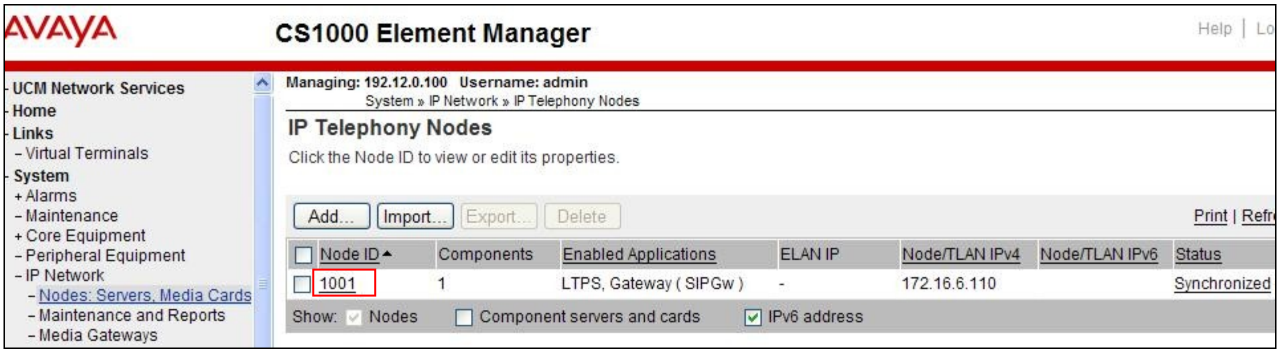

Whether the CS1000E is accessed directly or via System Manager, the Avaya Unified Communications Management **Elements** page will be used for configuration.

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

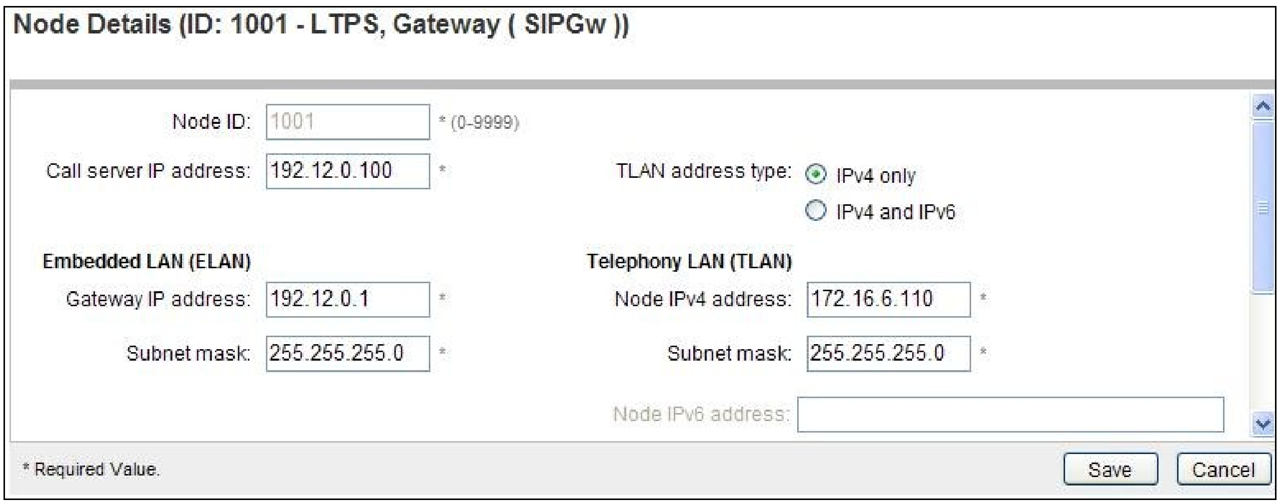



## 5.1. Node and Key IP Addresses

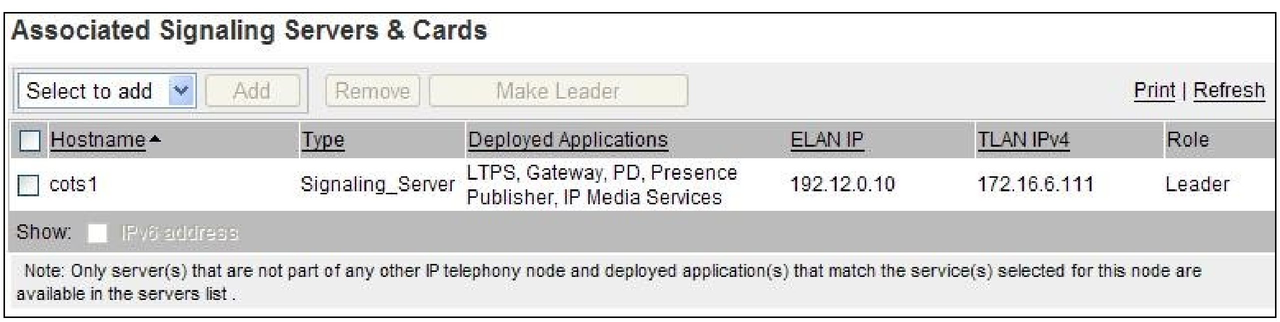
**Step 1** - Expand **System → IP Network** on the left panel and select **Nodes: Servers, Media Cards**. The **IP Telephony Nodes** page is displayed as shown below. Click "**<Node id>**" in the **Node ID** column to view details of the node. In the sample configuration, **Node ID "1001"** was used.



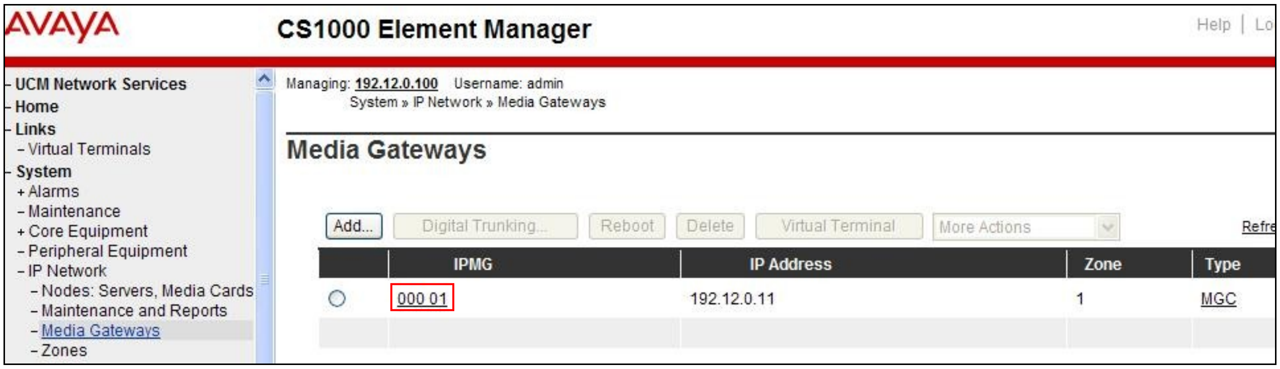

The **Node Details** screen is displayed with additional details as shown below. Under the **Node Details** heading at the top of the screen, make a note of the **TLAN Node IPV4 address.** In the sample screen below, the **Node IPV4 address** is **"**172.16.6.110**".** This IP address will be needed when configuring Session Manager with a SIP Entity for the CS1000E in **Section 6.4.1**.




The following screen shows the **Associated Signaling Servers & Cards** heading at the bottom of the screen, simply to document the configuration.

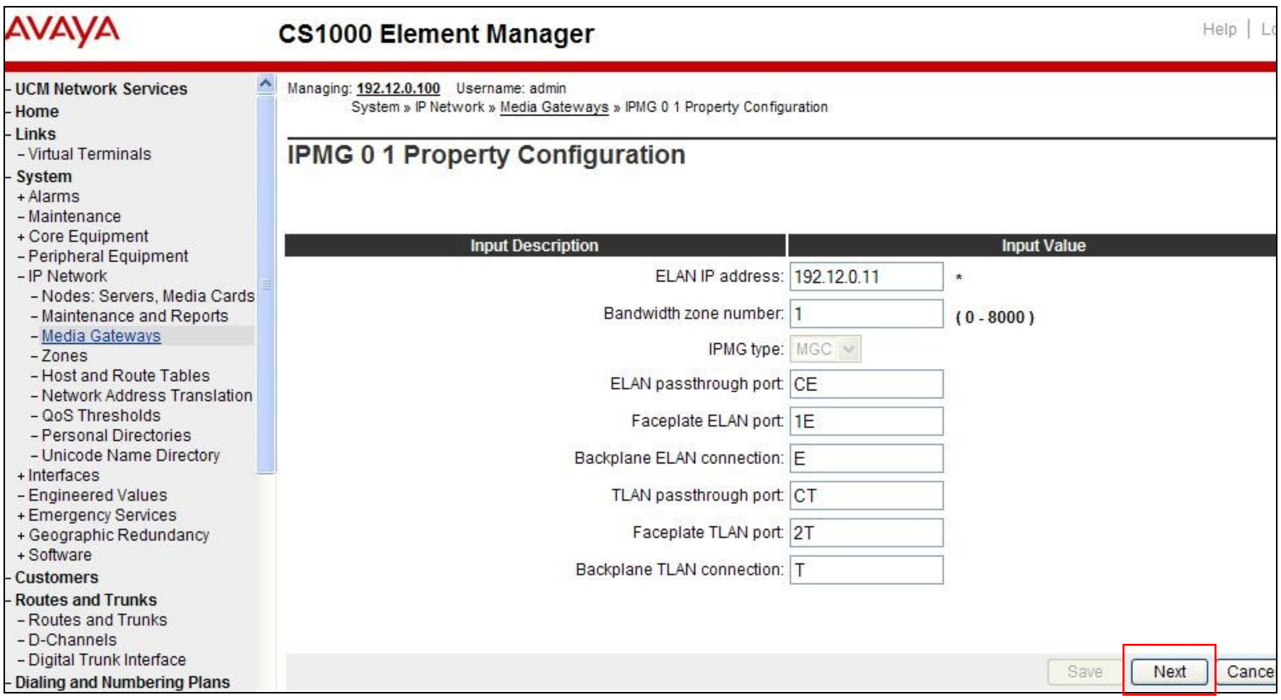



**Step 2** - Expand **System → IP Network** on the left panel and select **Media Gateways**. Click on the IPMG ID (e.g. **000 01**).

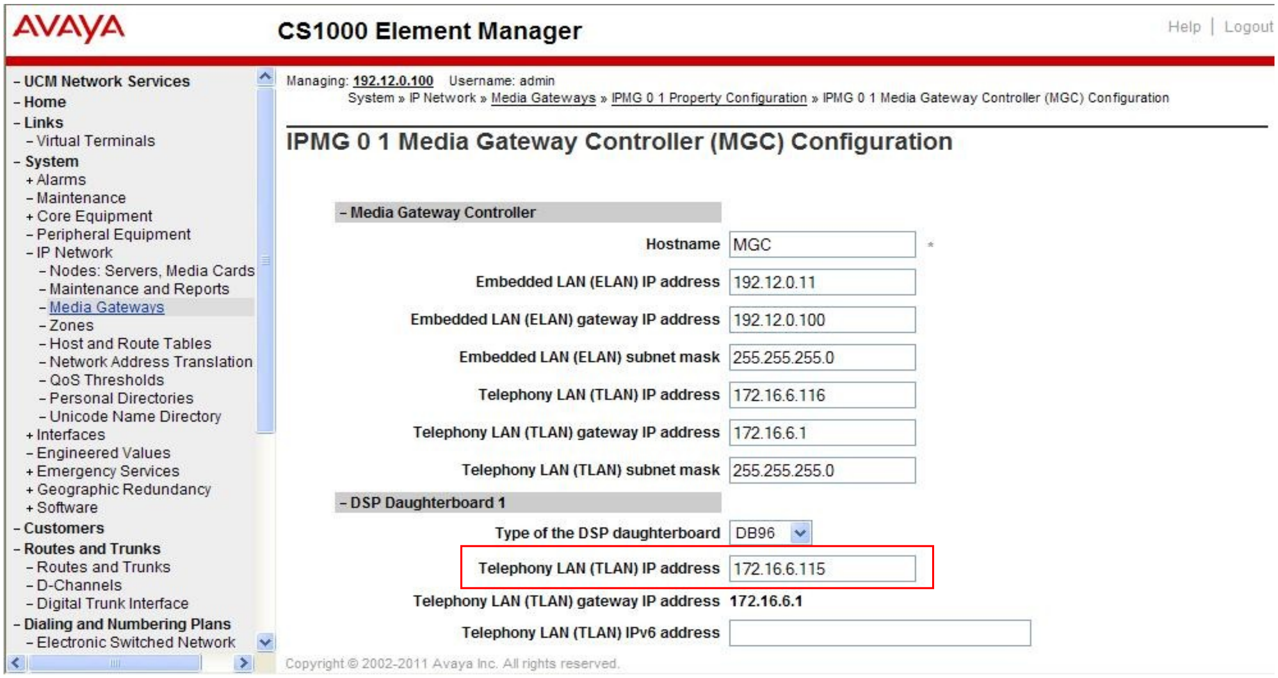



This will open the Property Configuration screen.

**Step 3** – Click on the Next button.




This will open the MGC Configuration screen. The **Telephony LAN (TLAN) IP Address** under the **DSP Daughterboard 1** heading will be the IP Address in the SDP portion of SIP messages, for calls requiring MGC resources. For example, for a call from an analog or digital telephone to PSTN, the IP Address in the SDP in the INVITE message that the CS1000E sends to Session Manager, and on to the Avaya Aura® SBC, will be 172.16.6.115 in the sample configuration. Note that the Avaya Aura® SBC will change this IP address to the Avaya Aura® SBC "outside" IP address before sending the INVITE on to the AT&T IP Flexible Reach service.

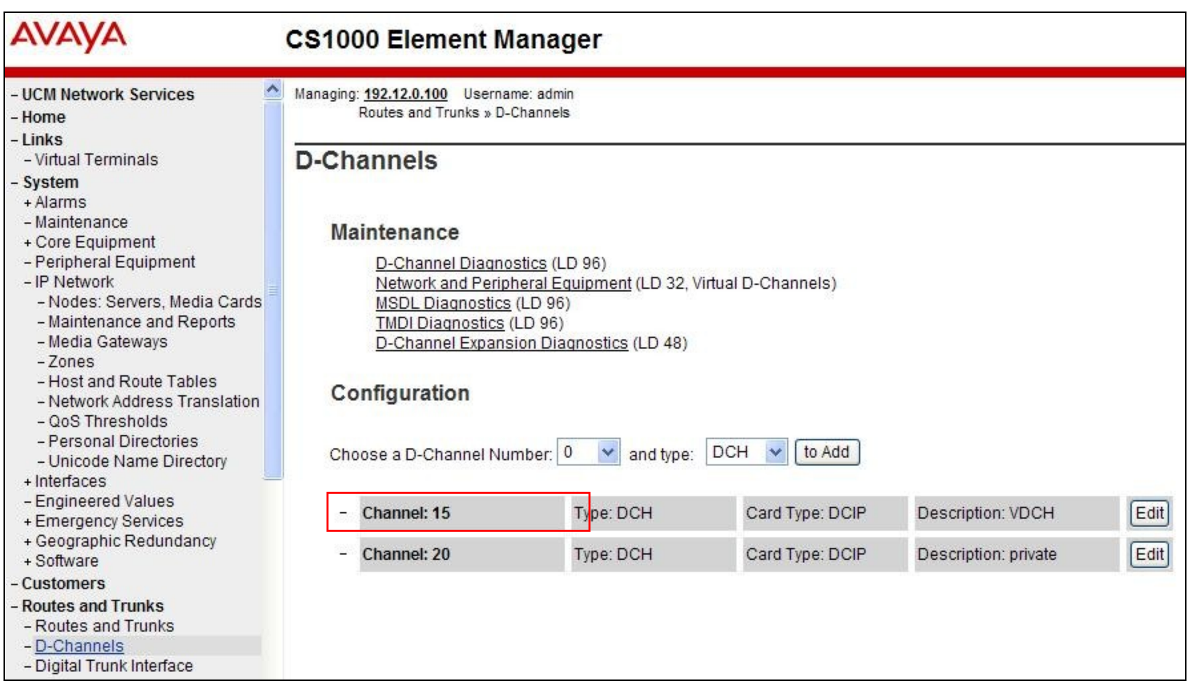



## 5.2. Virtual D-Channel, Routes and Trunks

Avaya Communication Server 1000E Call Server utilizes a virtual D-channel and associated Route and Trunks to communicate with the Signaling Server.

### 5.2.1 Virtual D-Channel Configuration

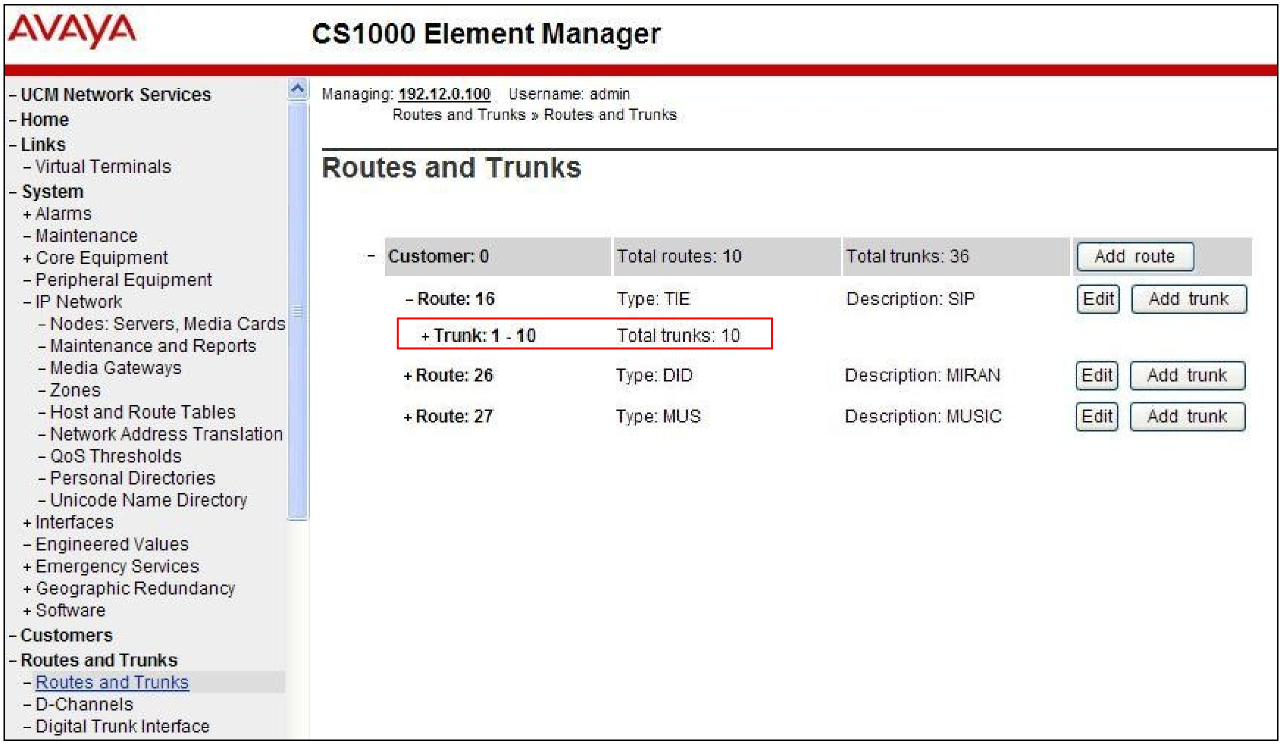
Expand **Routes and Trunks** on the left navigation panel and select **D-Channels**. In the sample configuration, virtual D-Channel 15 is associated with the Signaling Server.



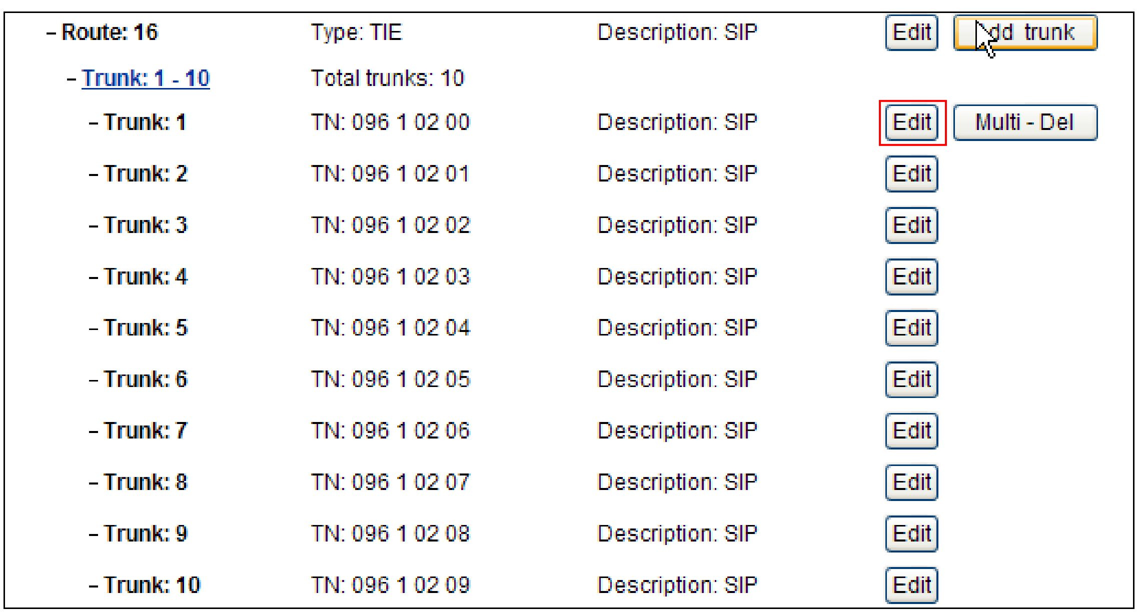
## 5.2.2 Routes and Trunks Configuration

In addition to configuring a virtual D-channel, a **Route** and associated **Trunks** must be configured.

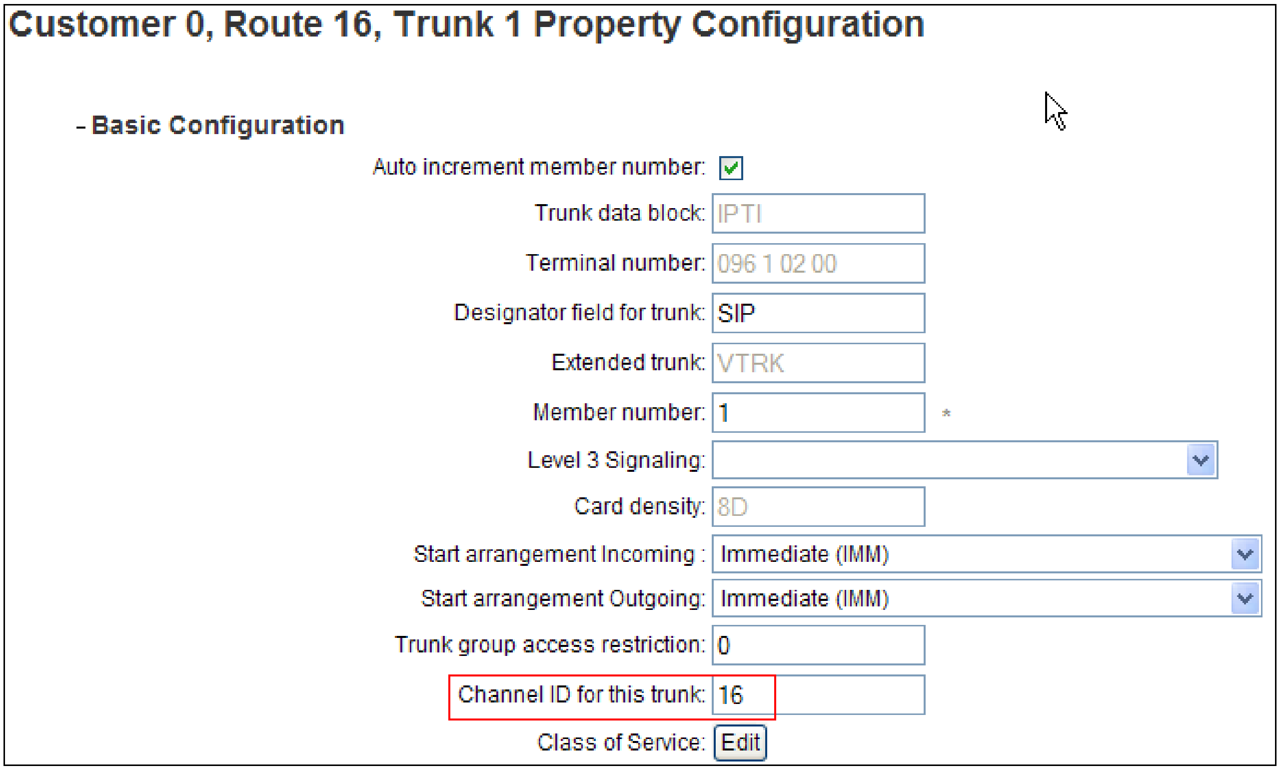
**Step 1** - Expand **Routes and Trunks** on the left navigation panel and expand the Customer number (e.g. **Customer 0**). In the example screen that follows, it can be observed that **Route 16** has 10 trunks in the sample configuration (**Trunk:1 – 10**).



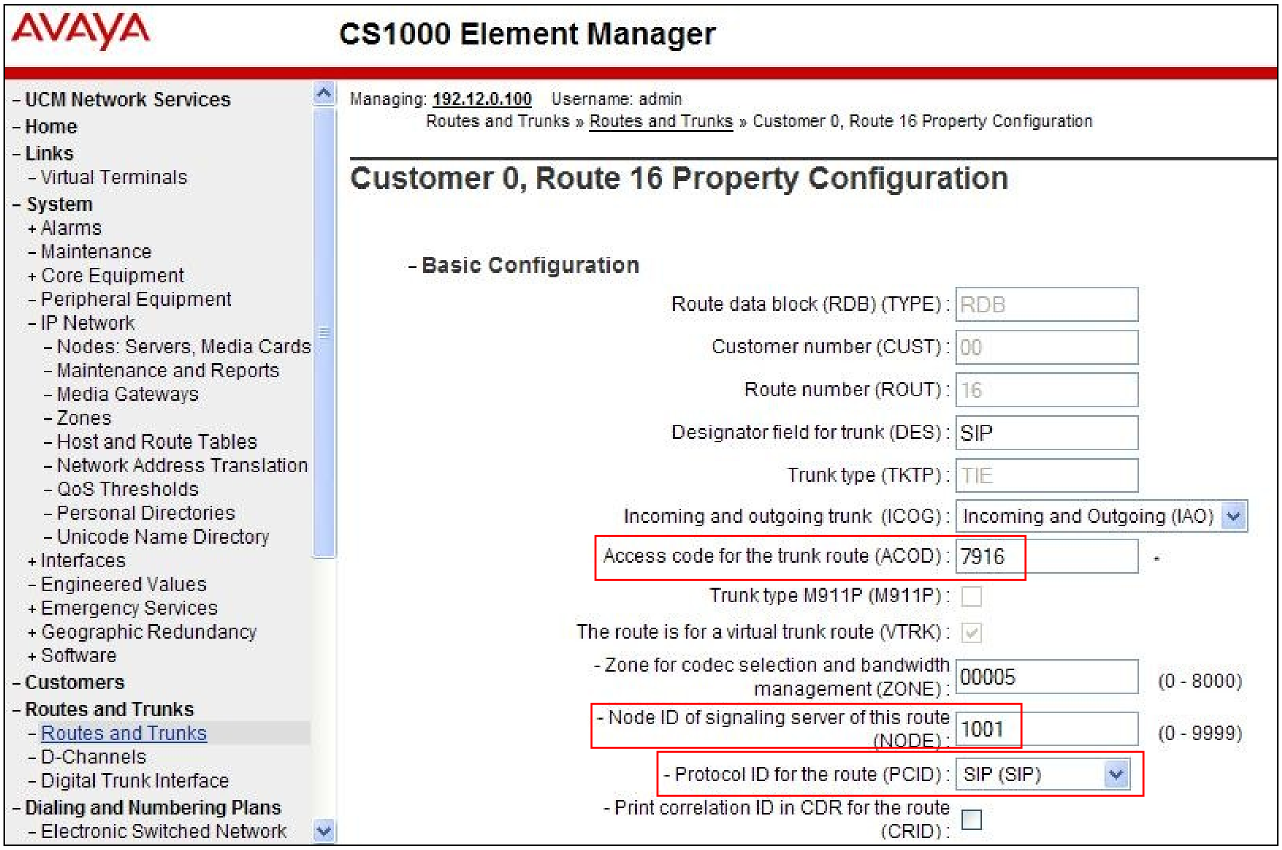

**Step 2** – Click on **Trunk:1-10** to display each trunk channel.



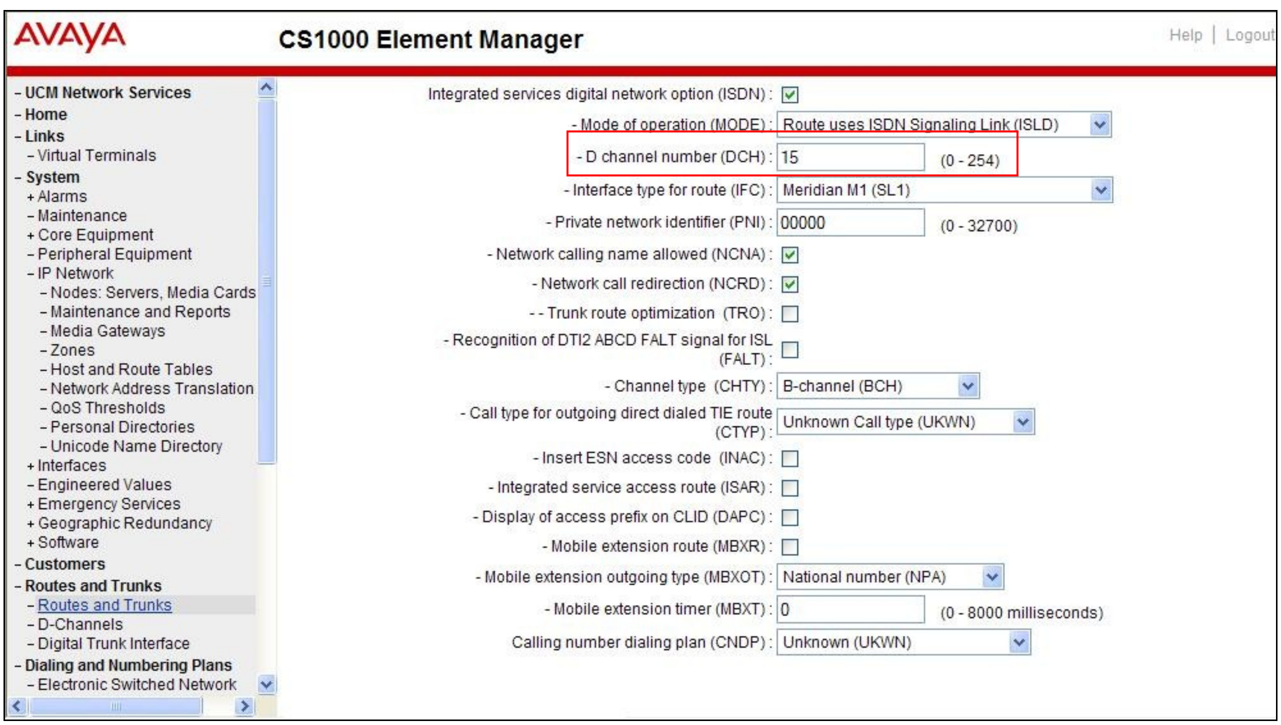
**Step 3** – Click on the **Edit** button for **Trunk: 1**. Each trunk that is defined must have a corresponding channel, but these trunk and channel numbers need not match. Because channels 1-15 were allocated by other provisioning, in the reference configuration Trunk **1** used Channel **16**. Therefore, each subsequent trunk that was provisioned used channel 16+1. For example, Trunk 9 shown in the table above will use channel 24 (16+8 = 24).



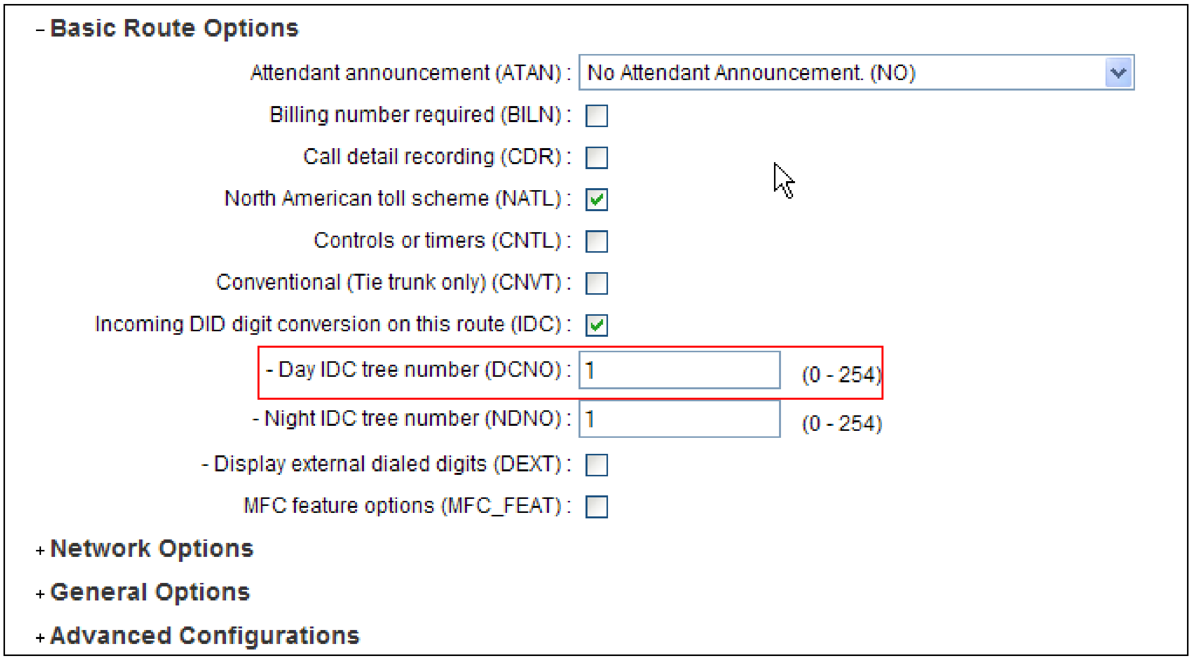

**Step 4** – Going back to the screen shown in **Step 1**, select the **Edit** button next to **Route 16** to verify the configuration, as shown below. Verify "**SIP (SIP)**" has been selected for **Protocol ID for the route (PCID)** field and the **Node ID of signaling server of this route (NODE)** matches the node shown in **Section 5.1**. 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 Avaya 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.




**Step 5** - Scrolling down, other parameters may be observed. The **D channel number (DCH)** field must match the D-Channel number shown in **Section 5.2.1**.



**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 5.5**.

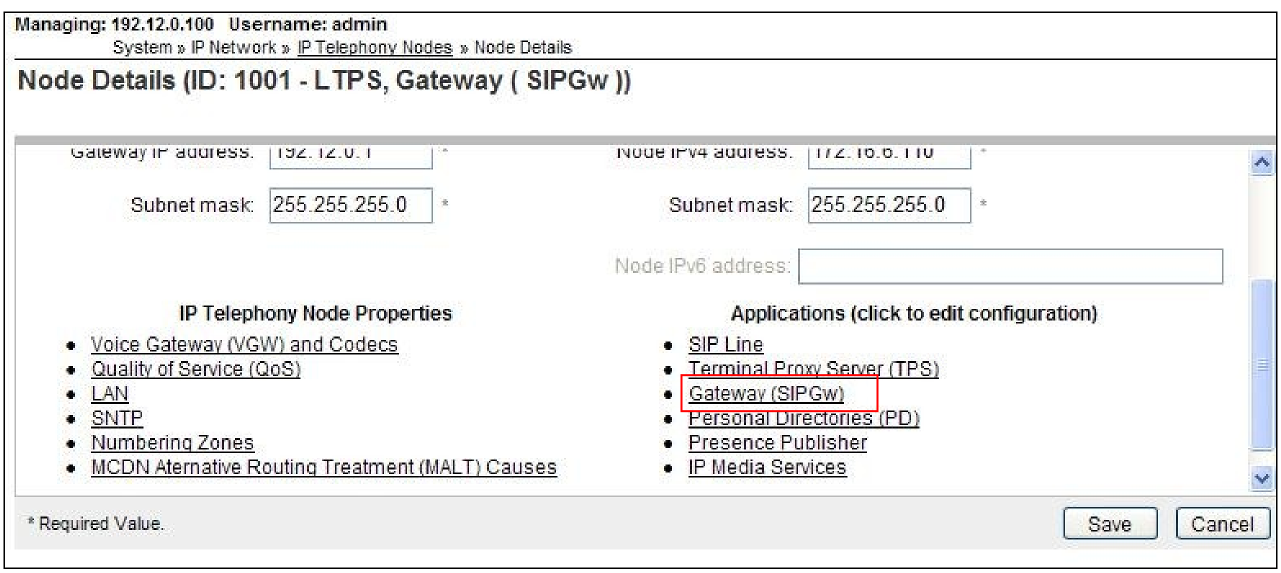



## 5.3. SIP Trunk to Session Manager

**Step 1** - Expand **System → IP Network → Nodes: Servers, Media Cards**.

**Step 2** - Select **Node ID 1001** as shown in **Step 2** of **Section 5.1** to edit configuration settings for the configured node.

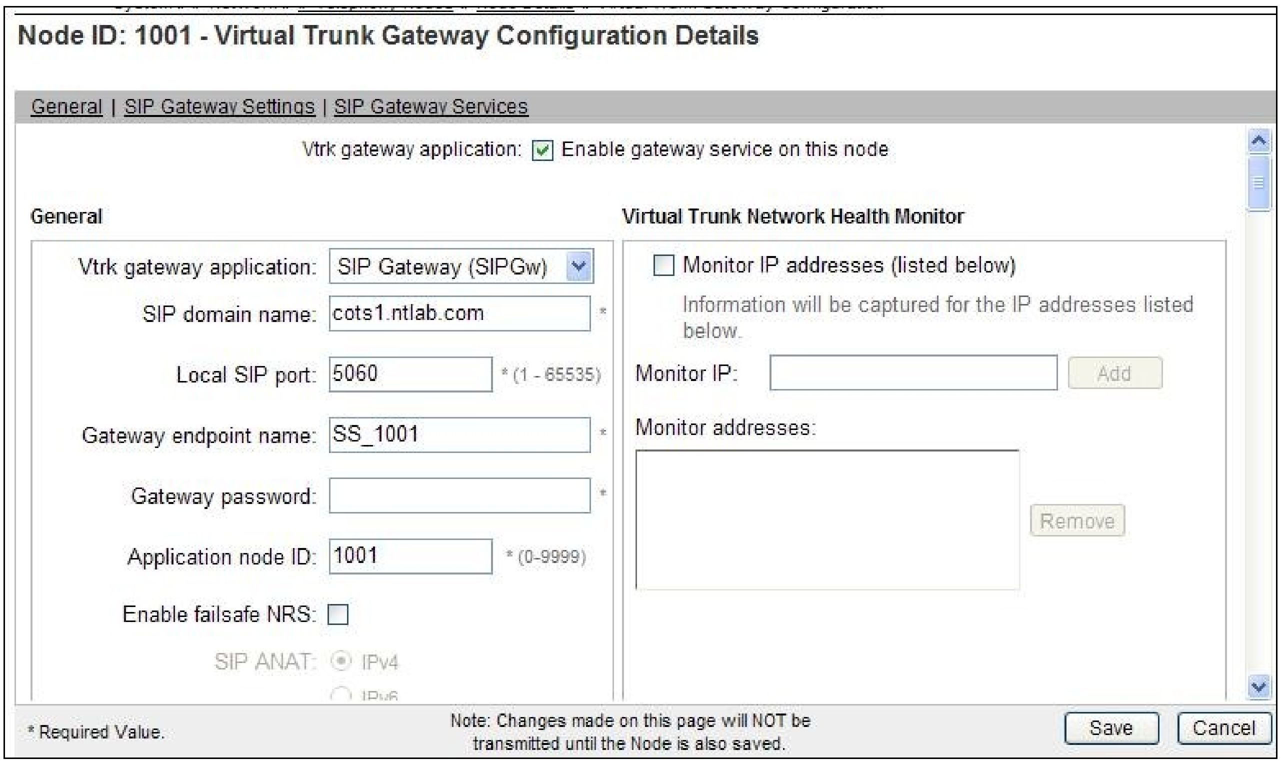
**Step 3** - Using the scroll bar on the right side of the screen, navigate to the **Applications** section on the screen and select the **Gateway (SIPGw)** link to view or edit the SIP Gateway configuration.



**Step 4** - 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.
- **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 5.1**.

The values defined for the sample configuration are shown below.



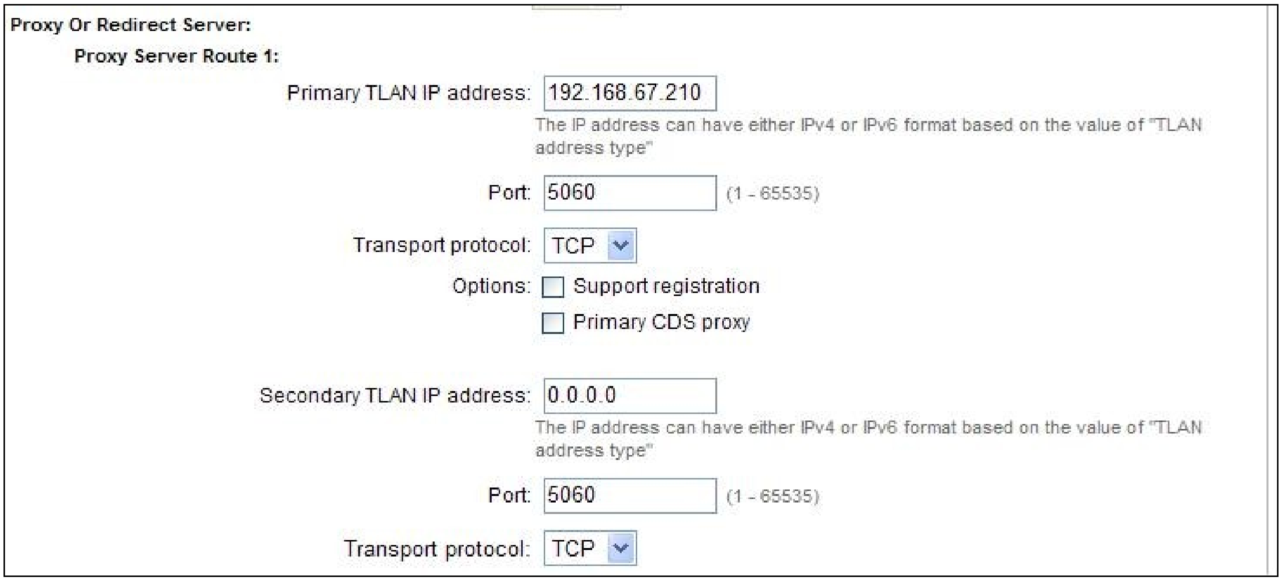

**Step 5** - Scroll down to the section: **SIP Gateway Settings → Proxy or Redirect Server**

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, "**192.168.67.210**" was used.
- **Port:** Enter "**5060**"
- **Transport protocol:** Select "**TCP**"

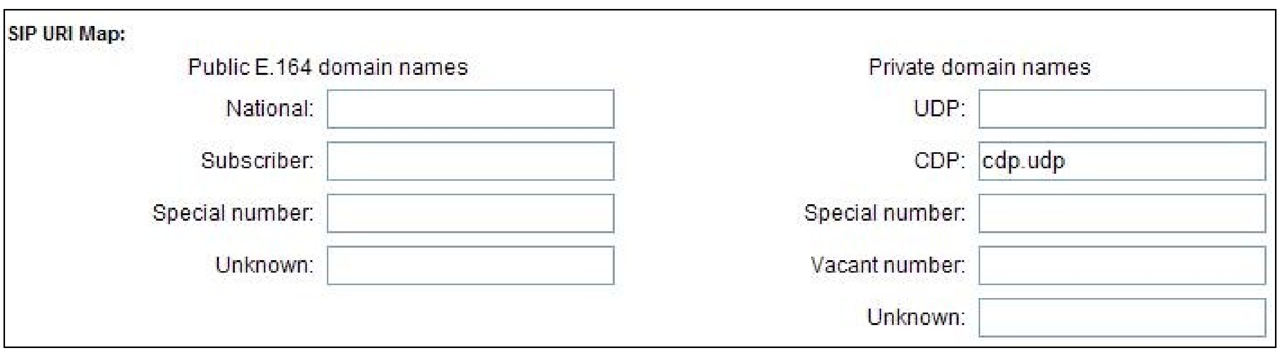
**Note**: TCP was used for the reference configuration. However, Avaya best practices recommends the use of TLS in production environments. For more information on configuring the CS1000E to use TLS, see [8].

**Note** - The Secondary TLAN IP address was not used.




**Step 6** - Scroll down and repeat these steps for the **Proxy Server Route 2** (not shown).

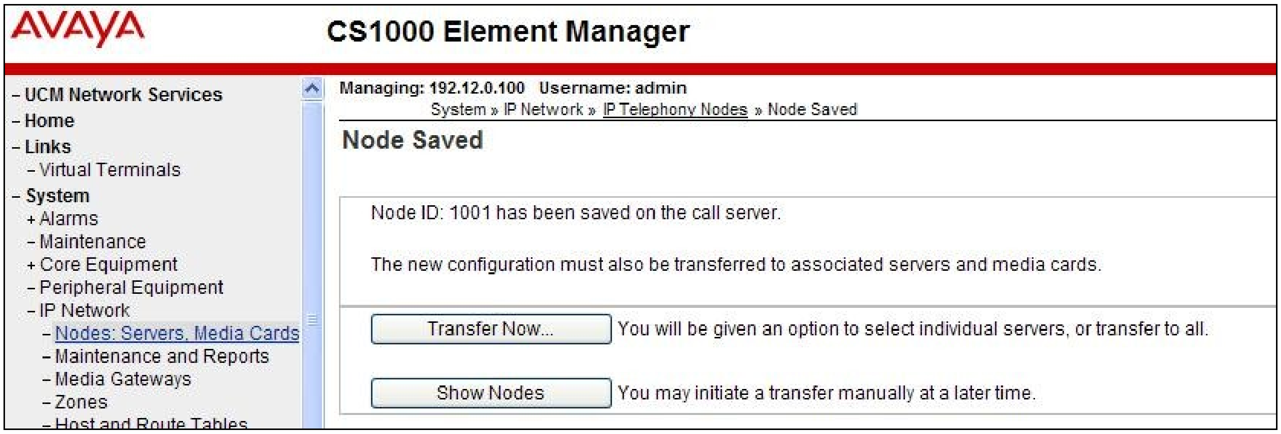
**Step 7** - Scroll down to the **SIP URI Map** section. The values defined for the sample configuration are shown below. The Avaya CS1000E will put the "string" entered in the **SIP URI Map** in the "phone-context=<string>" parameter in SIP headers such as the P-Asserted-Identity. If the CDP: value is configured to blank, the CS1000E will omit the "phone-context=" in the SIP header altogether.



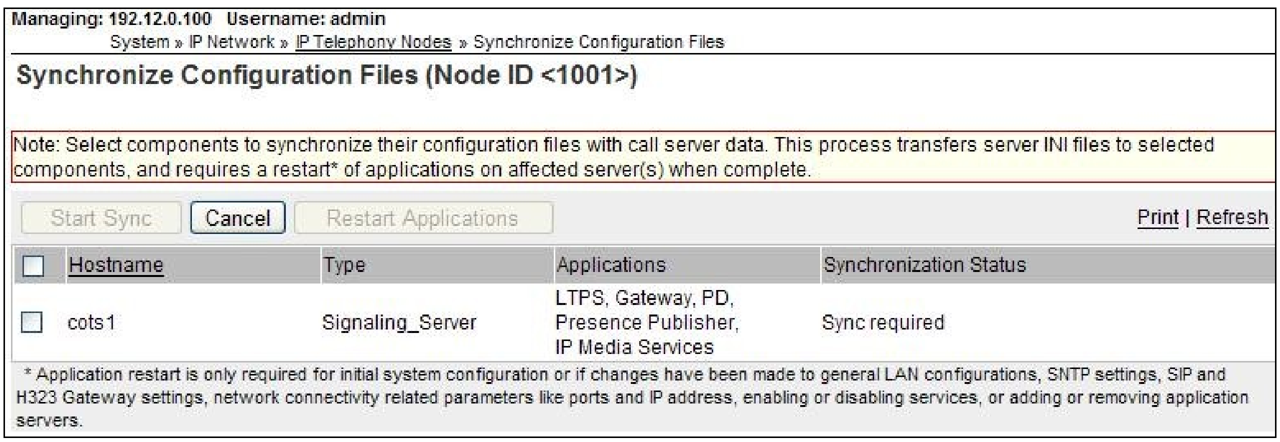

**Step 8** - Scroll to the bottom of the page and click **Save** (not shown) to save SIP Gateway configuration settings. This will return the interface to the **Node Details** screen.

**Step 9** - Click **Save** on the **Node Details** screen (not shown).

**Step 10** - Select **Transfer Now** on the **Node Saved** page as shown below.

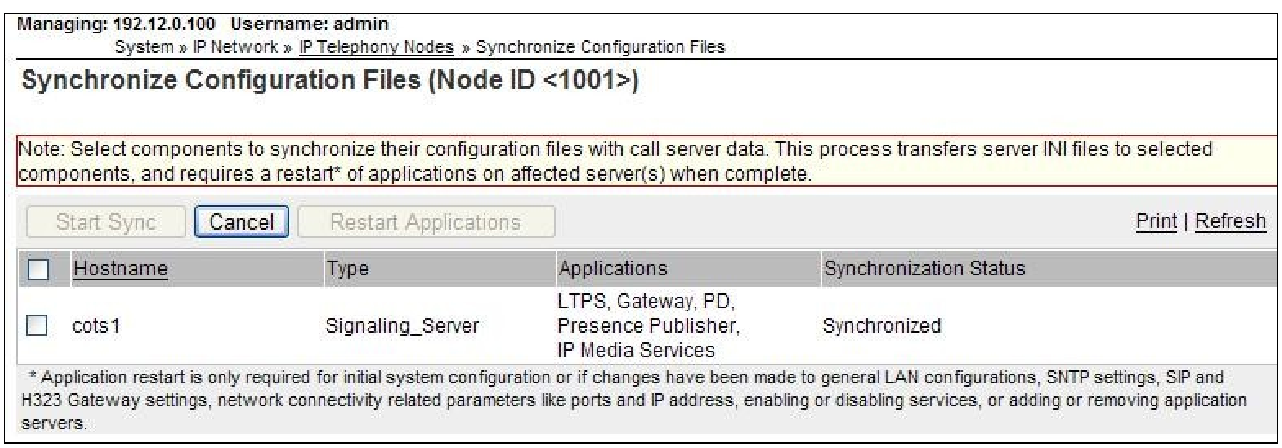



Once the transfer is complete, the **Synchronize Configuration Files (Node ID <id>)** page is displayed.



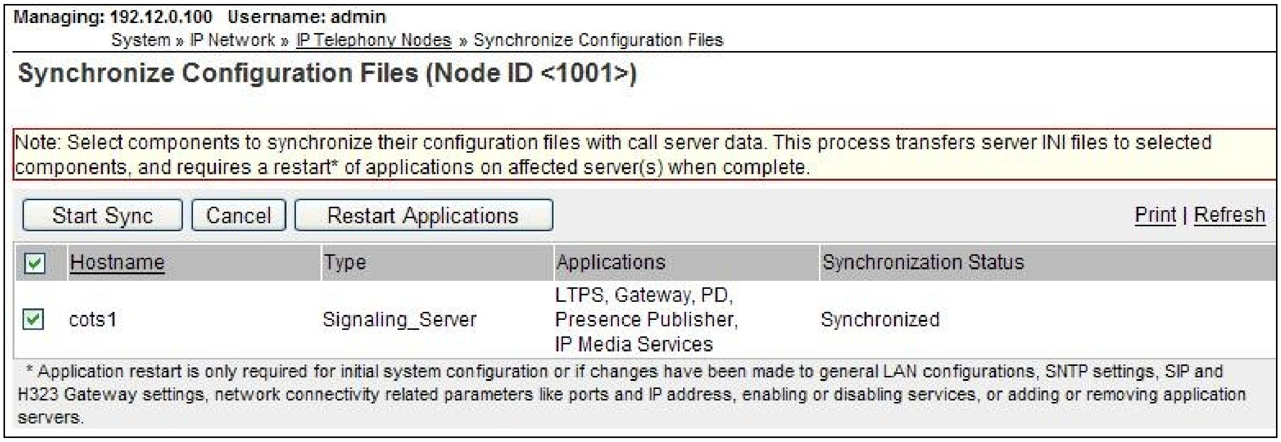

**Step 11** - Enter associated with the appropriate Hostname (e.g. **cots1**) and click **Start Sync.**

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



**Step 12** - After synchronization completes, click on the **Refresh** button in the right hand corner, enter associated with the appropriate Hostname (e.g. cots1), and click **Restart Applications**.

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

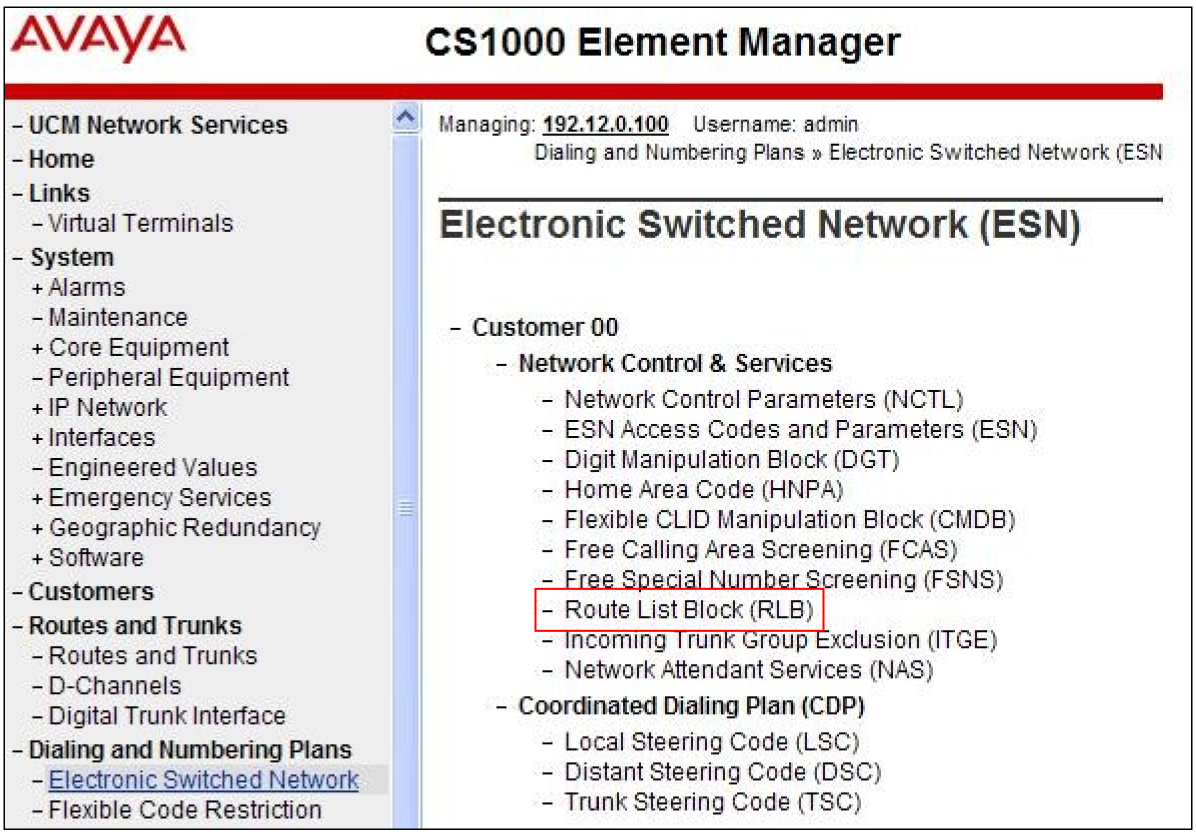


## 5.4. Routing of Outbound Dialed Numbers to Session Manager

This section provides the configuration of the routing used in the reference configuration for routing calls over the SIP Trunk between Avaya Communication Server 1000E and Session Manager for calls destined for the AT&T IP Flexible Reach service. The routing defined in this section is simply an example and not intended to be prescriptive. The example will focus on the configuration enabling a CS1000E telephone user to dial 9-1-732-xxx-xxxx to reach a PSTN telephone. Other routing policies may be appropriate for different customer networks.

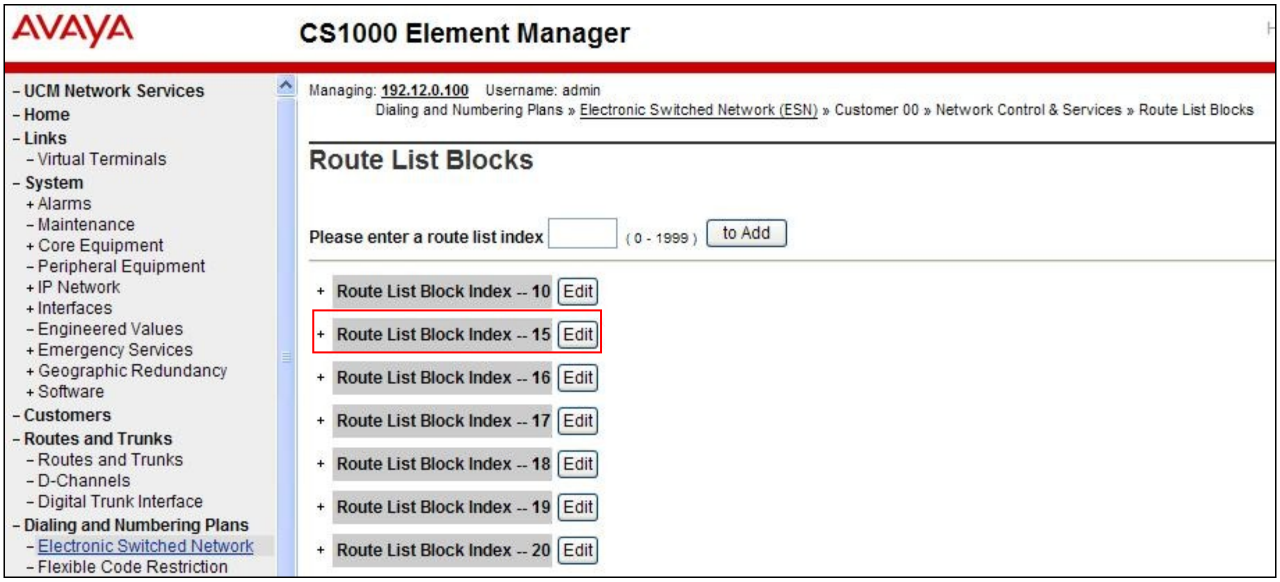
### 5.4.1 Route List Block

**Step 1** - Expand **Dialing and Numbering Plans** on the left navigational panel and select **Electronic Switched Network.** Select **Route List Block (RLB)** on the **Electronic Switched Network (ESN)** page as shown below.



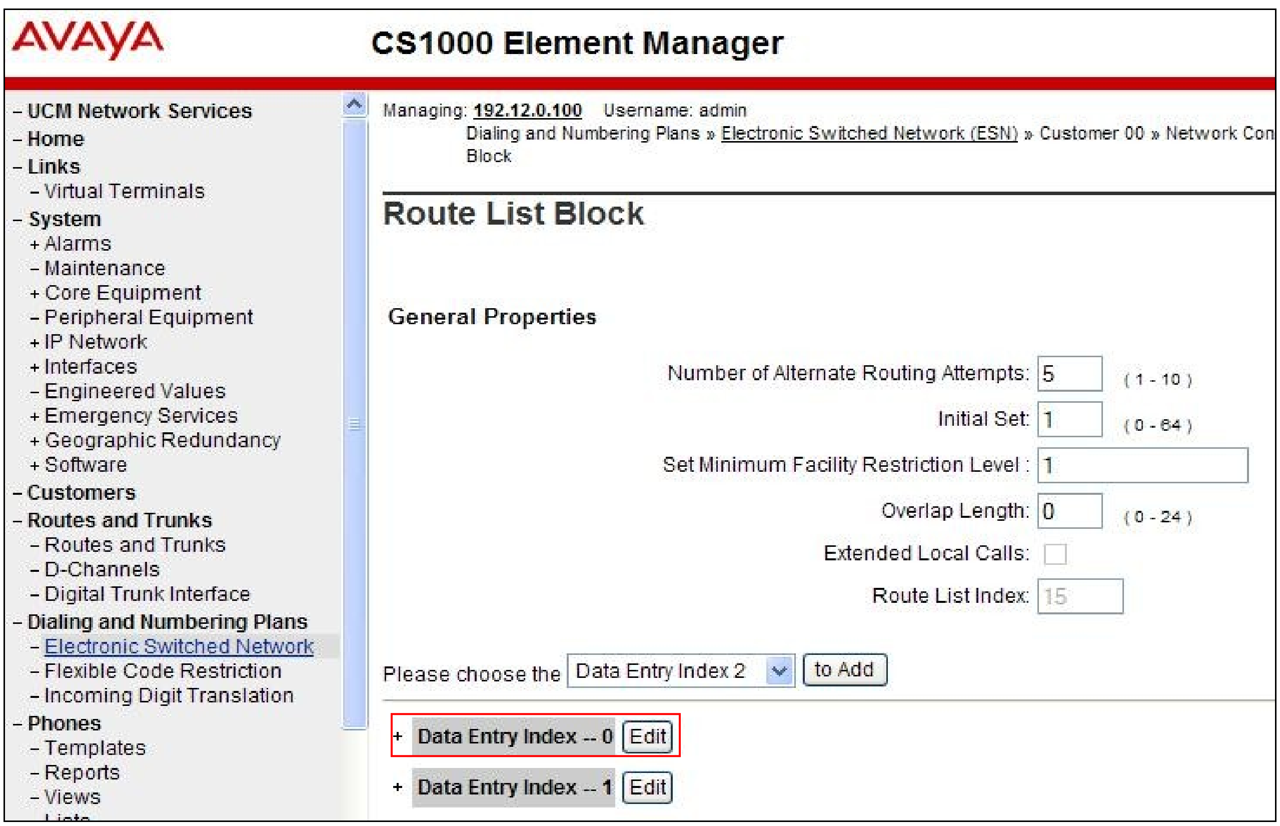

The **Route List Blocks** screen is displayed.

**Step 2** - Enter an available route list index number in the **Please enter a route list index** field and click **to Add**, or edit an existing entry by clicking the corresponding **Edit** button. In the sample configuration, route list block index **15** is used.




The Route List Block screen will open.

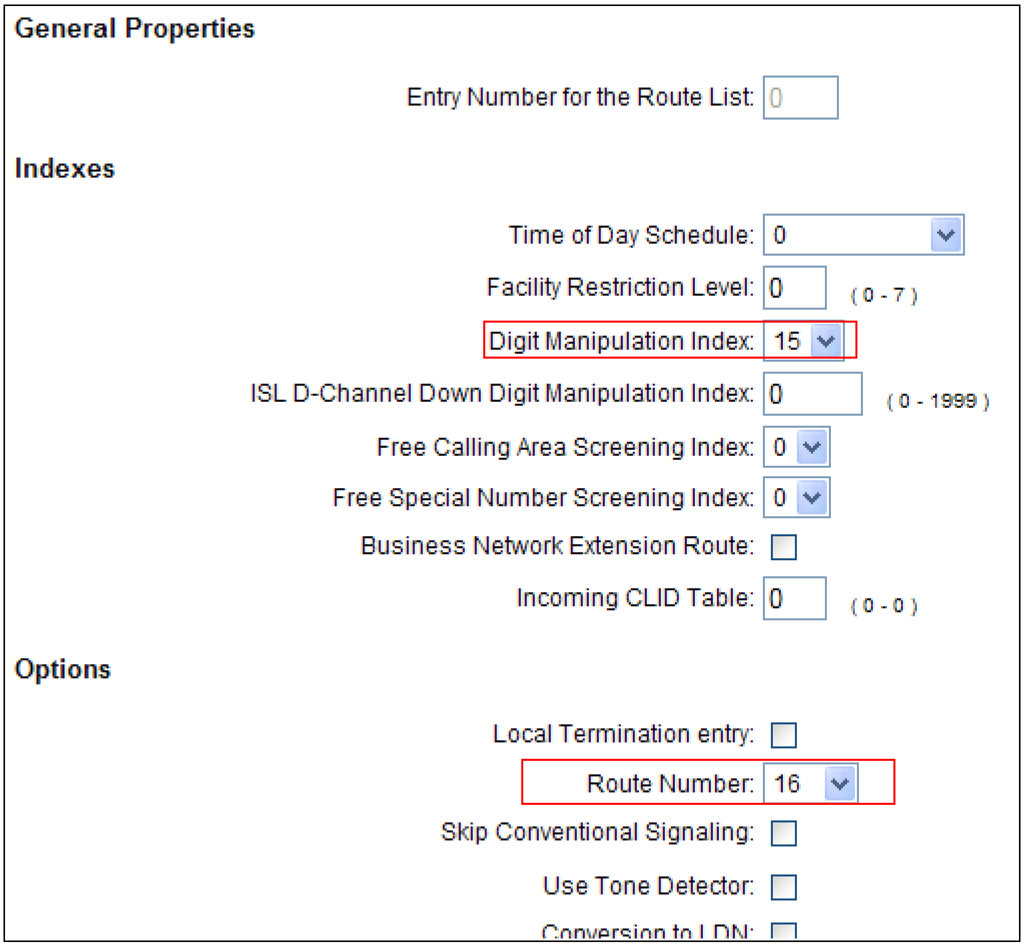
**Step 3** - If adding a new route list index, scroll down to the **Options** area of the screen. If editing an existing route list block index, select the **Edit** button next to the appropriate **Data Entry Index** as shown below (e.g. **0**).




The **Data Entry of a Route List Block** screen will open.

**Step 4** – Scroll down to **Digit Manipulation Index** and select **15** (see **Section 5.4.2**).

**Step 5** - Scroll down to the **Options** section and select a **“<Route id>”** in the **Route Number** drop down menu. In the sample configuration route number **16** was used. Default values may be retained for remaining fields as shown below.



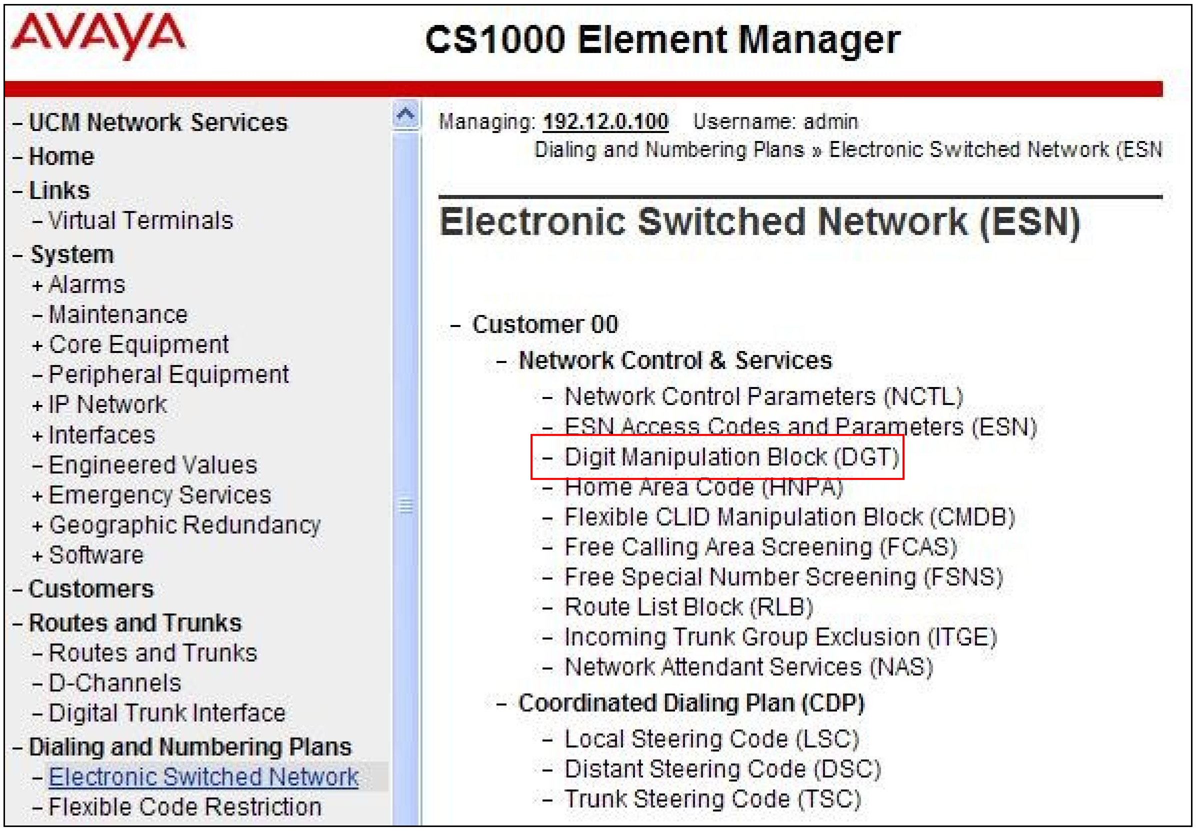

**Step 6** - Click **Submit** (not shown) to save the Route List Block definitions.

In the reference configuration Route list block 15 uses Digit Manipulation 15 to keep the called number unchanged (see below), and Route 16 to send calls to Session Manager.

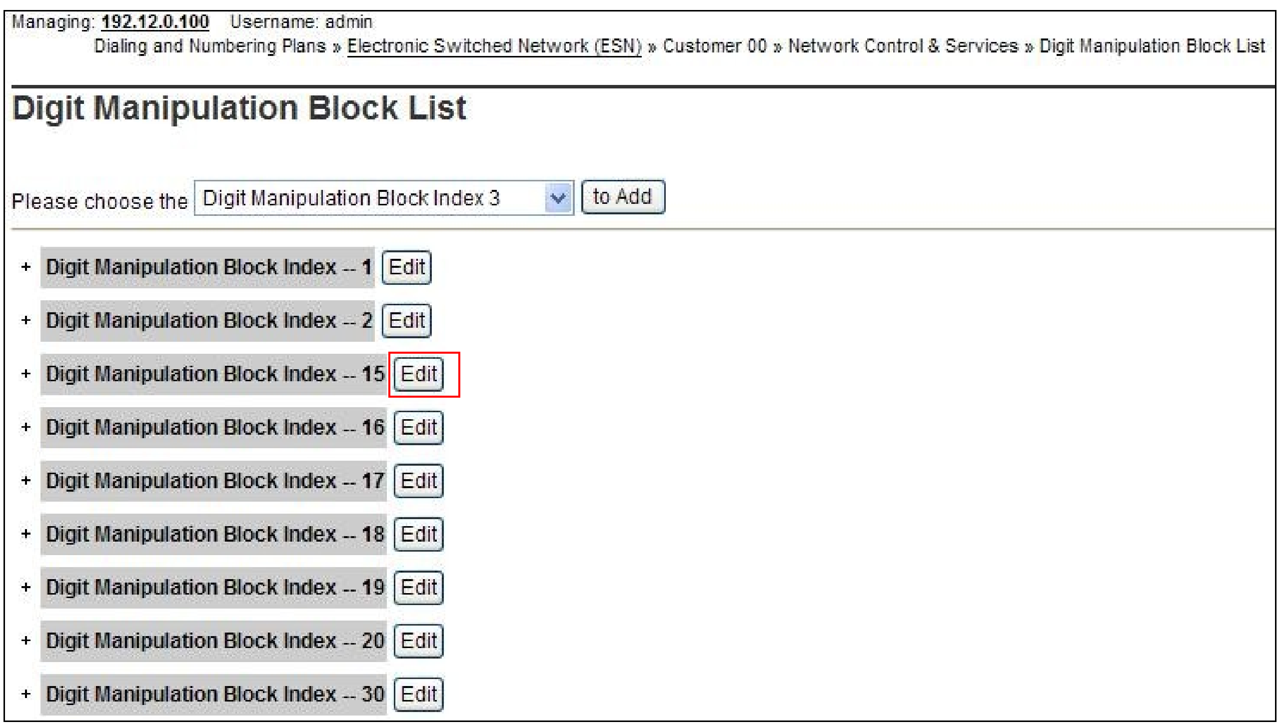
### 5.4.2 Digit Manipulation Block

The Digit Manipulation Block (DGT) is used to modify the outbound called digit string.

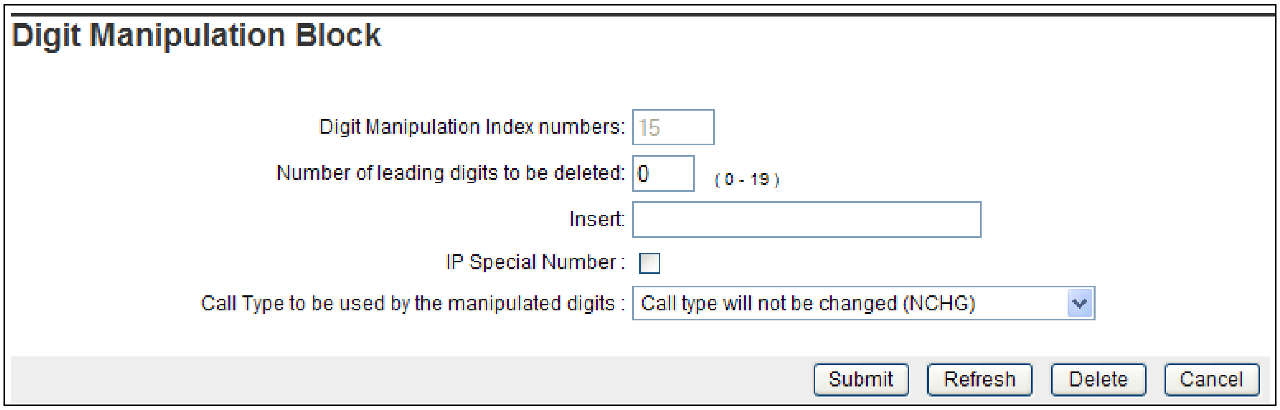
**Step 1** - Expand **Dialing and Numbering Plans** on the left navigational panel and select **Electronic Switched Network.** Select **Digit Manipulation Block (DGT)** as shown below.




**Step 2** – Add a new Digit Manipulation Block if required. In the reference configuration Digit Manipulation Block **15** was used. Click on **Edit.**



**Step 3** – Set **Number of leading digits to be deleted** to **0** (zero). Set **Call Type to be used by the manipulation digits** to **Call type will not be changed (NCHG)**.




**Step 4** – Click on **Submit**.

### 5.4.3 NARS Access Code

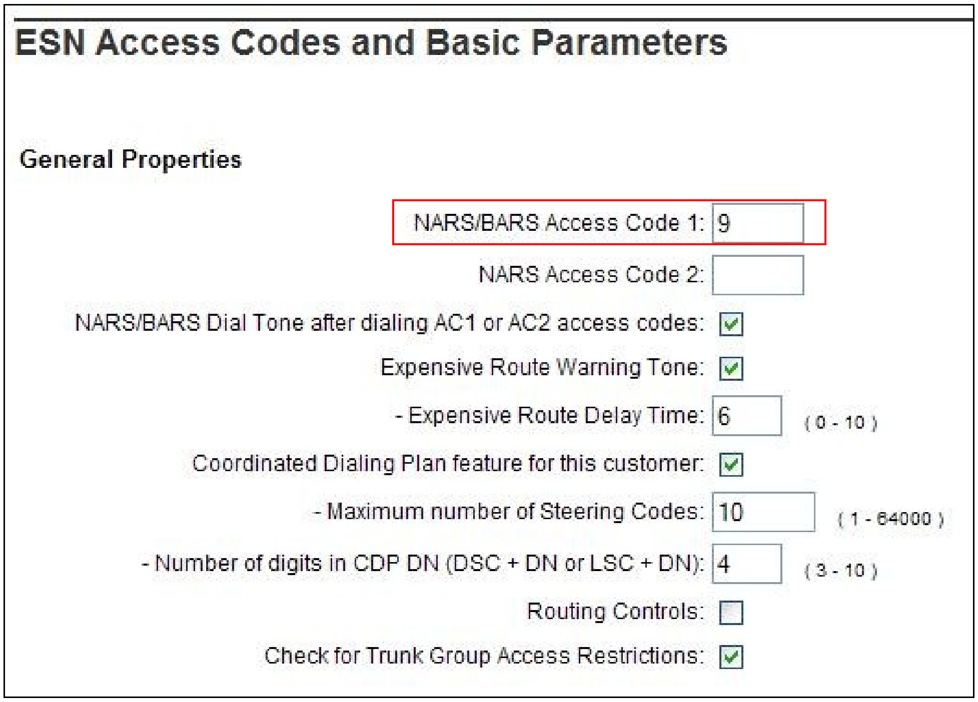
This section defines the access code for off-net dialing (e.g. calls to PSTN).

**Step 1** - Expand **Dialing and Numbering Plans** on the left navigational panel and select **Electronic Switched Network.**

**Step 2 -** Select **ESN Access Codes and Parameters (ESN).** Although not repeated below, this link can be observed in the first screen in **Section 5.4.1**.

**Step 3** - In the **NARS/BARS Access Code 1** field, enter the number the user will dial before the target PSTN number. In the sample configuration, the single digit "**9**" was used.

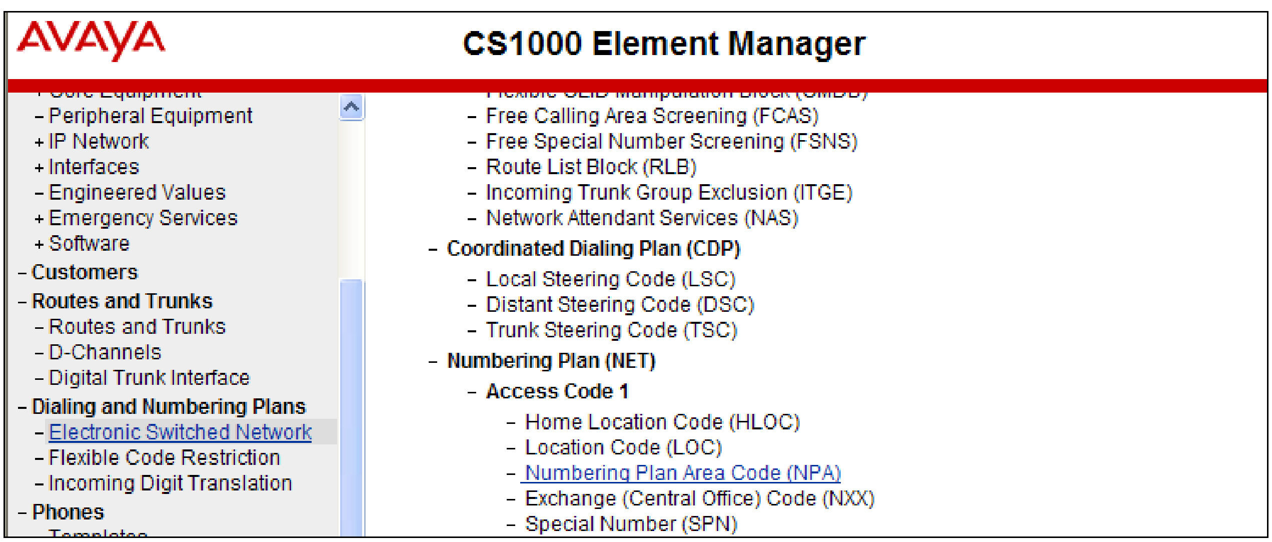
**Step 4** - Click on **Submit** (not shown).



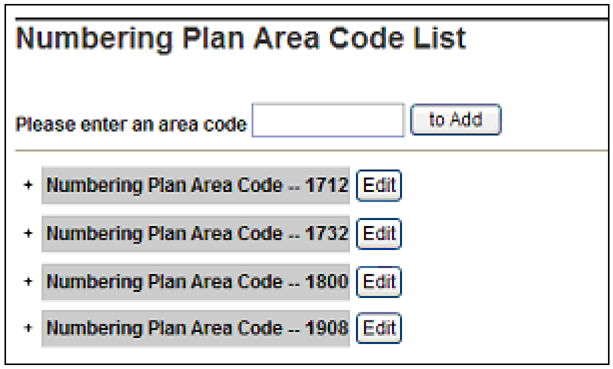

## 5.4.4 Numbering Plan Area Codes

This section defines the various **Numbering Plan Area Code (**NPA) used to access PSTN (e.g. **1732**).

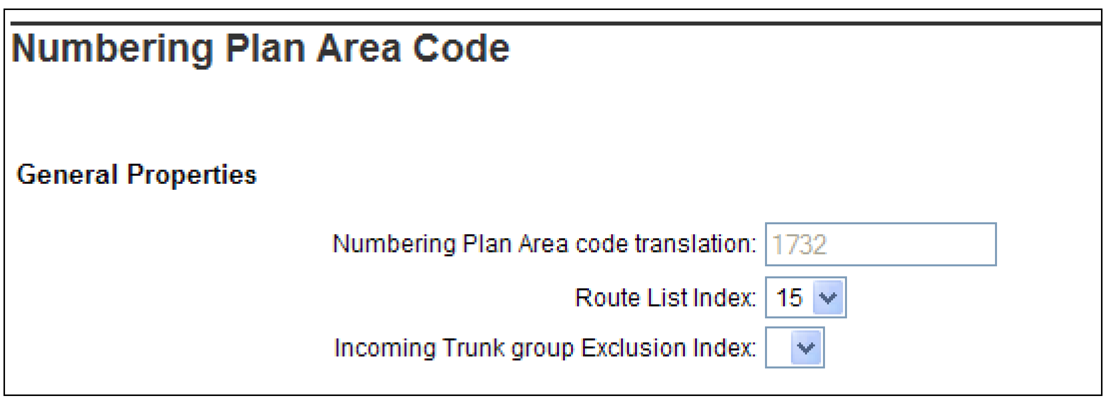
**Step 1** - Expand **Dialing and Numbering Plans** on the left navigational panel and select **Electronic Switched Network.** Scroll down and select **Numbering Plan Area Code (NPA)** under the appropriate access code heading. In the sample configuration, this is **Access Code 1**, as shown in below.



**Step 2** - Add a new NPA by entering it in the **Please enter an area code** box and click **to Add** or click **Edit** to view or change an NPA that has been previously configured. In the screen below, it can be observed that various dial strings such as **1732**, **1800** and **1908** are configured.




**Step 3** - In the screen below, the entry for "**1732**" is displayed. In the Route List Index, "**15**" is selected to use the route list associated with the SIP Trunk to Session Manager (as defined in **Section 5.4.1**, **Step 2**). Default parameters may be retained for other parameters. Repeat this procedure for the dial strings associated with other numbering plan area codes that should route to the SIP Trunk to Session Manager.



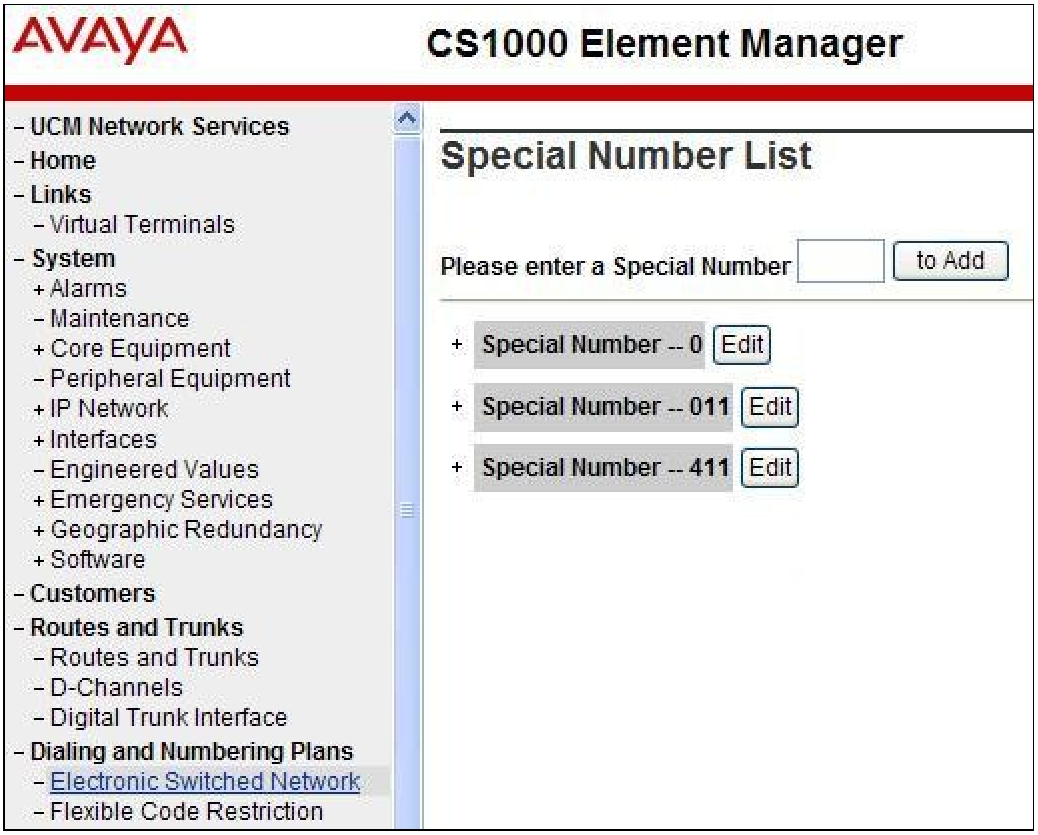

### 5.4.5 Other Special Numbers to Route to Session Manager

In the testing associated with these Application Notes, non-emergency service numbers such as **n11**, and **011** international calls were also routed to Session Manager and ultimately to the AT&T IP Flexible Reach service. Although not intended to be prescriptive, one approach to such routing is summarized in this section.

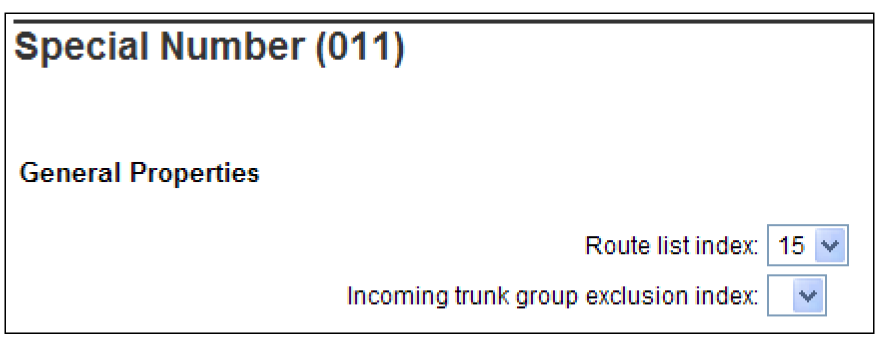
**Step 1** - Expand **Dialing and Numbering Plans** on the left navigational panel and select **Electronic Switched Network.**

**Step 2 -** Scroll down and select **Special Number (SPN)** under the appropriate **Access Code** heading (e.g. **1** as shown in **Section 5.4.3, Step 3**).

**Step 3** - Add a new number by entering it in the **Please enter a Special Number** box and click **to Add** or click **Edit** to view or change a special number that has been previously configured. In the screen below, it can be observed that various dial strings such as 0, 011, and x11 calls are listed.




**Step 4** – To modify an entry click on "**Edit**". In each case, **Route list index** "**15**" has been selected in the same manner as shown for the NPAs in the prior section.




**Step 4** - Click on **Submit** (not shown).

### 5.4.6 Summary

In summary, to have CS1000E route a PSTN call for 1732xxxxxxx via SIP trunk to Session Manager:

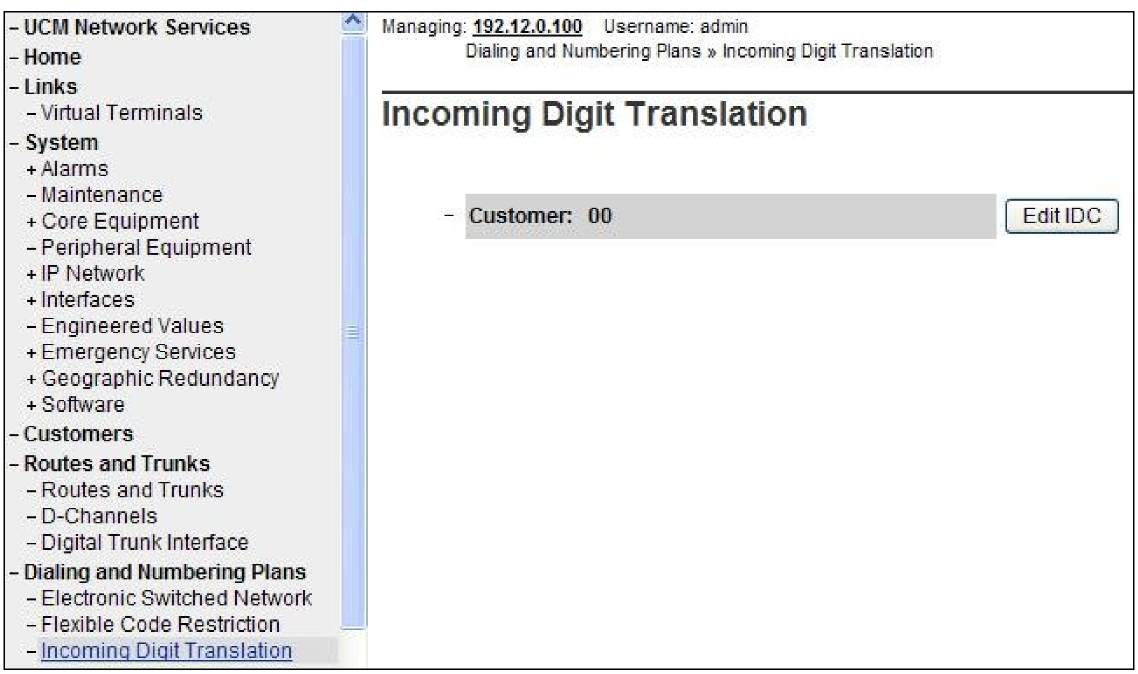
- Routes & Trunks (**Section 5.2.2**)
    - Customer 0
    - Route 16 = SIP trunk
- Route List Block (**Section 5.4.1**)
    - Data 0
        - Route = 16
        - Digit Manipulation = 15
- Digit Manipulation Block 15 (**Section 5.4.2**)
    - Delete = 0
    - Type = NCHG
- ESN Access Codes and Parameters (ESN) (**Section 5.4.3**)
    - NARS/BARS Access Code 1 = 9
- Numbering Plan Area Code NPA (**Section 5.4.4**)
    - 1732xxxxxxx
    - Route list Index = 15
    - Digit Manipulation Block 15
        - Delete = 0
        - Type = NCHG

## 5.5. Routing of Inbound Numbers to CS1000E

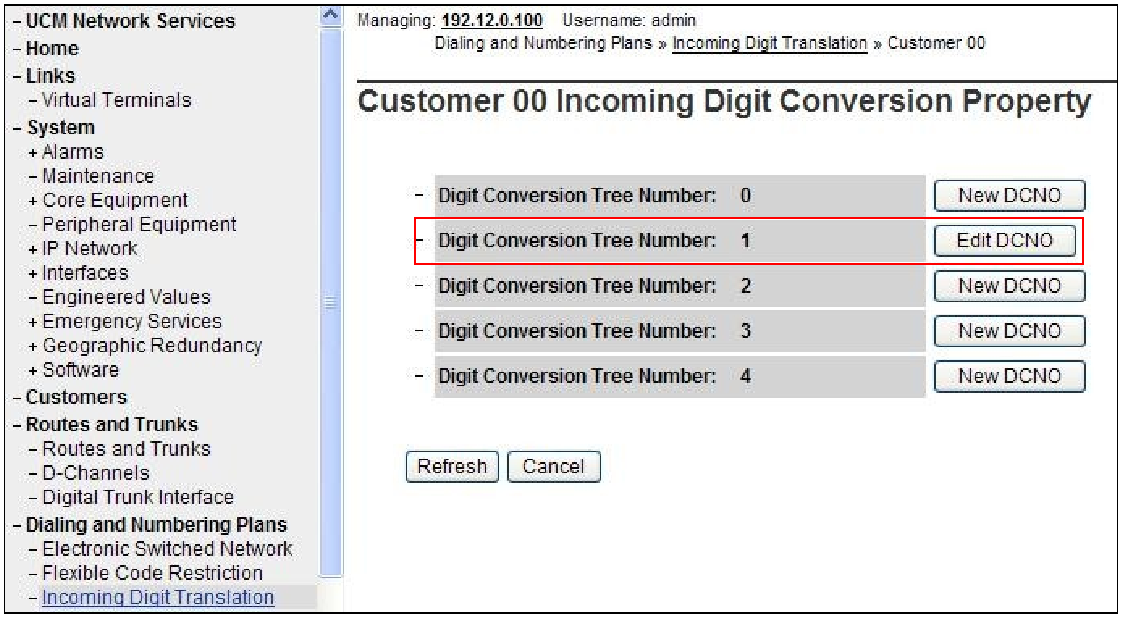
Calls from PSTN will dial AT&T IP Flexible Reach DID numbers to reach stations on the CS1000E. These DID numbers are converted to the associated extensions by the CS1000E Incoming Digit Translation (IDT) table.

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

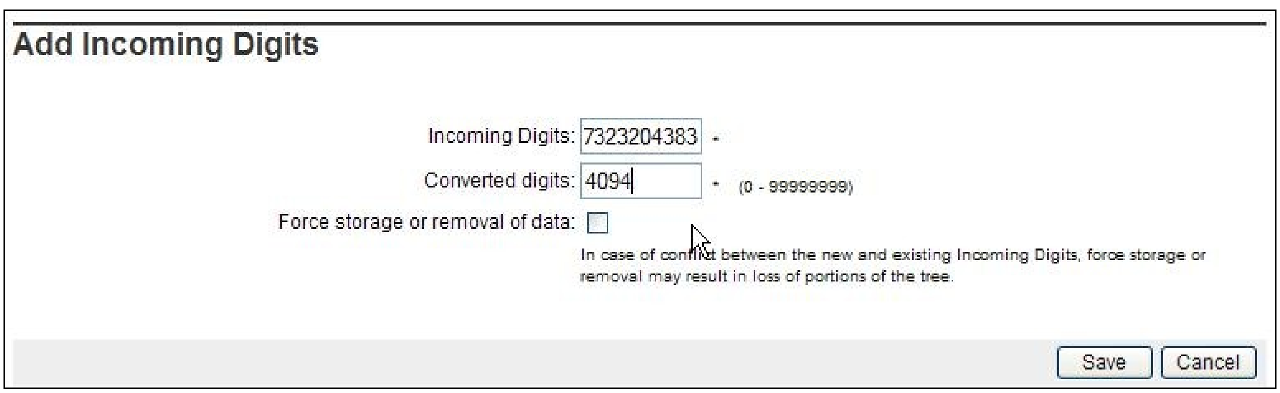
**Step 2 –** Select the appropriate **Customer ID** (**00** in the reference configuration) and click on **Edit IDC**.



**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 Routes and Trunks provisioning shown in **Section 5.2.2**, **Step 6**.




**Step 4** – The IDC Tree form will open. Click on the **Add** button. In the **Incoming Digits** field enter an AT&T IP Flexible Reach DID (e.g. **7323204383**). In the **Converted Digits** field enter the associated CS1000E extension (e.g. **4094**). Click on **Save**.



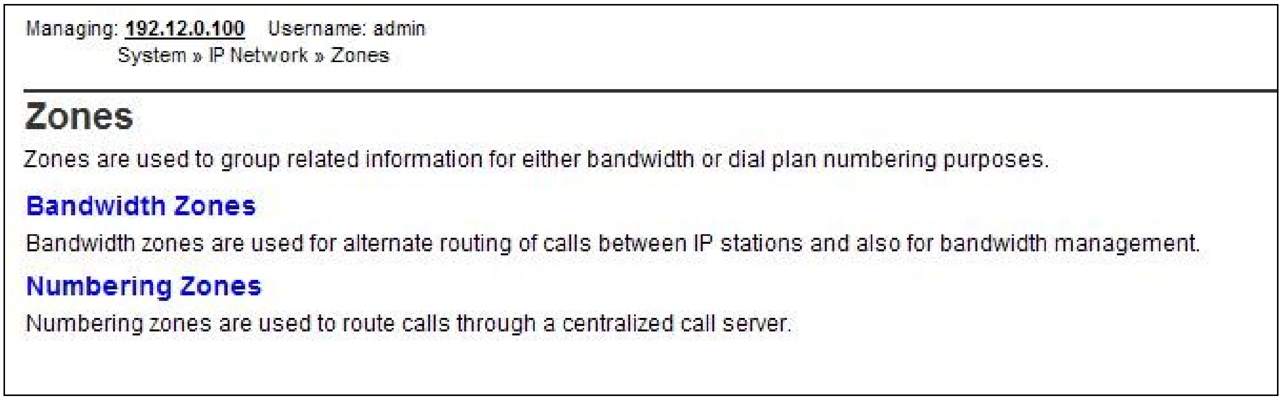

**Step 5** – Repeat **Step 4** for all associated AT&T IP Flexible Reach DIDs and extensions.

**Note** – This method should not be used to redirect DIDs for PSTN access to the Call Pilot access extension. The procedures described in Section 7.2.9 cover this scenario.

## 5.6. Zones

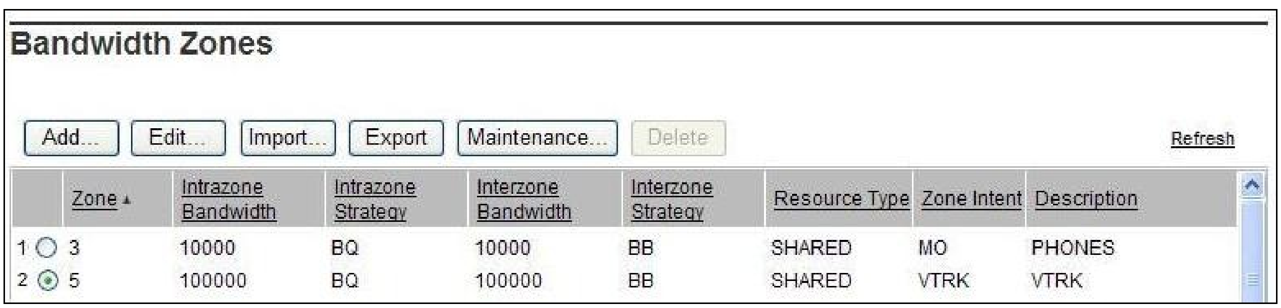
Zone configuration can be used to control codec selection and for bandwidth management.

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



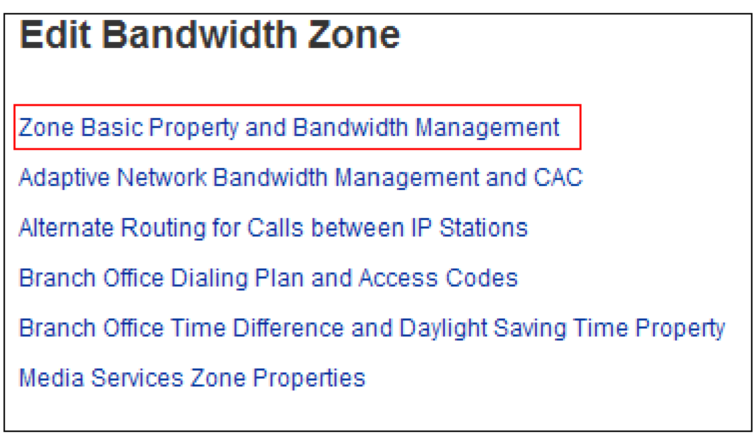

**Step 2** - Select **Bandwidth Zones**. In the sample lab configuration, two zones are configured as shown below. In production environments, it is likely that more zones will be required.

**Step 3** - Select the zone associated with the virtual trunk to Session Manager (e.g. item **2**, zone **5**) and click **Edit** as shown below.

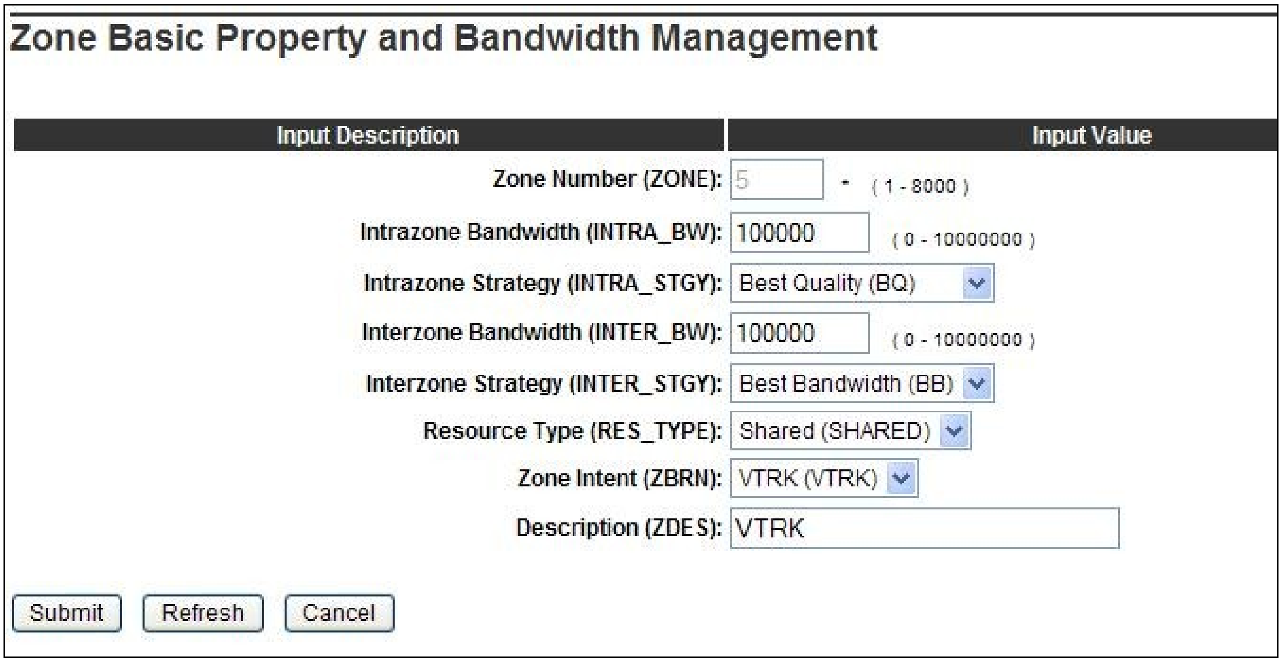



| | Zone | Intrazone Bandwidth | Intrazone Strategy | Interzone Bandwidth | Interzone Strategy | Resource Type | Zone Intent | Description |
|---|---|---|---|---|---|---|---|---|
| 1 | 3 | 10000 | BQ | 10000 | BB | SHARED | MO | PHONES |
| 2 | 5 | 100000 | BQ | 100000 | BB | SHARED | VTRK | VTRK |

**Step 4** - In the resultant screen shown below, select **Zone Basic Property and Bandwidth Management**.



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 for calls with the AT&T IP Flexible Reach service.




## 5.7. Codec Parameters.

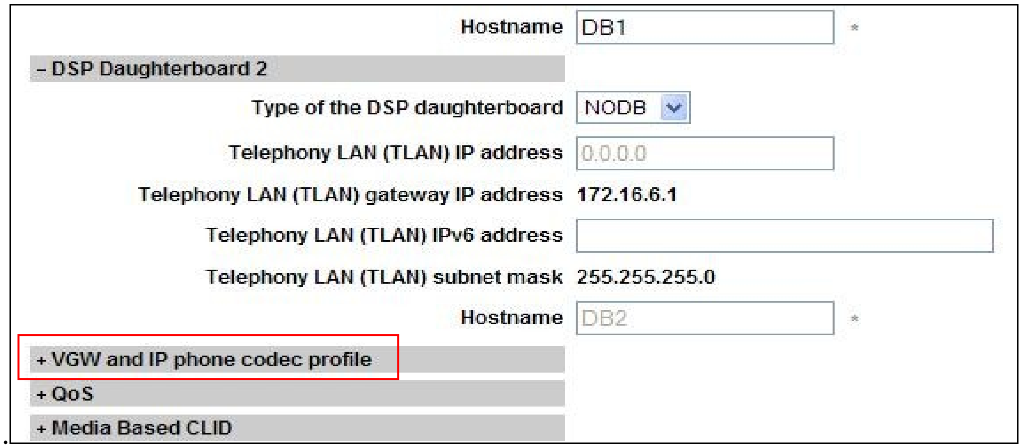
The following section describes how to set codec preferences as well as setting Packet Interval (PTIME) values. Note that the CS1000E always specifies G.711mu regardless of the additional selected codecs. Codecs are defined in the **Media Gateway** (for analog and digital phones) and in the **IP Telephony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

### 5.7.1 Media Gateway Codec Configuration

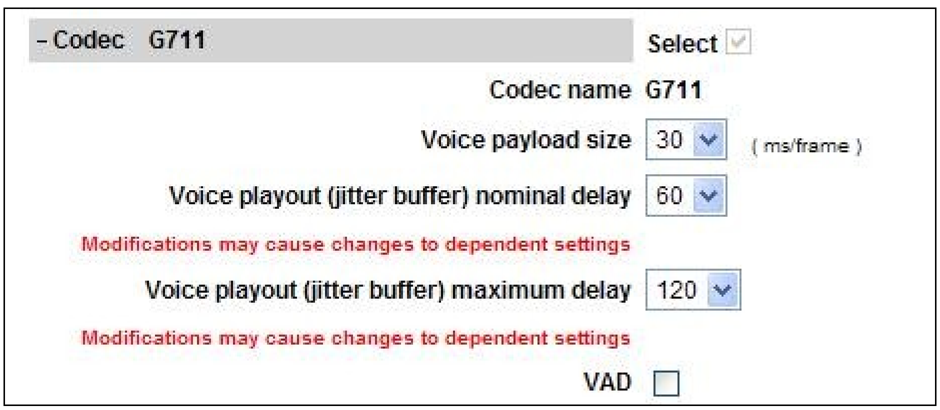
**Step 1** - Expand **System → IP Network** on the left panel and select **Media Gateways**.   Select the appropriate media gateway (e.g. **000 01** as shown in **Section 5.1**, **Step 2**).

**Step 2** - , The **Property Configuration** screen will open as shown in **Section 5.1**, **Step 3**. Click on "**Next**".

**Step 3** - Scroll down and click on **VGW and IP phone codec profile**.



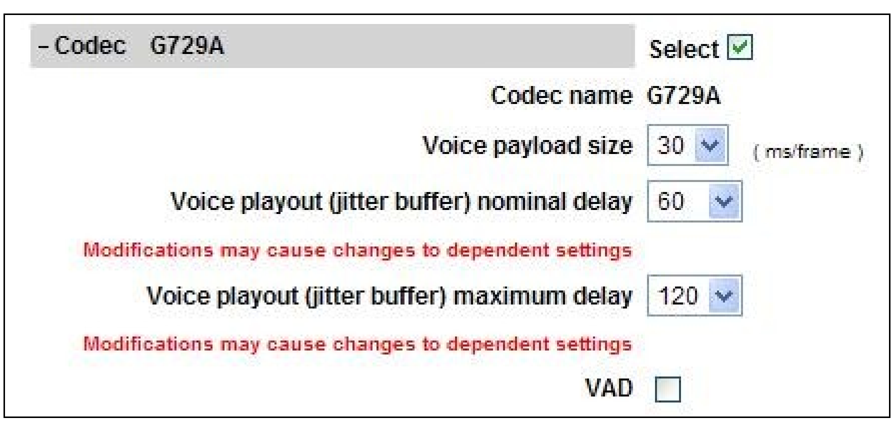

**Step 4** - The **VGW and IP phone codec profile** section will expand. Scroll down, click on and expand the **Codec G711** field. Note that the "Select" box is checked by default. Set the **Voice payload size** (PTIME) to **30**.



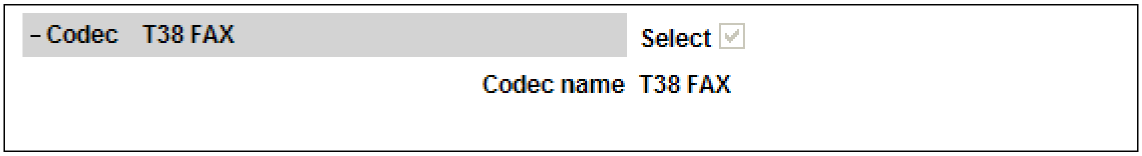

**Step 5** – Scroll down, click on and expand the **Codec G727A** field. Check the selection box and set the **Voice payload size** (PTIME) to **30.**

**Note** – Although not set in the reference configuration, annexB=yes may be enabled by selecting the **VAD** (Voice Activity Detection) box.

.

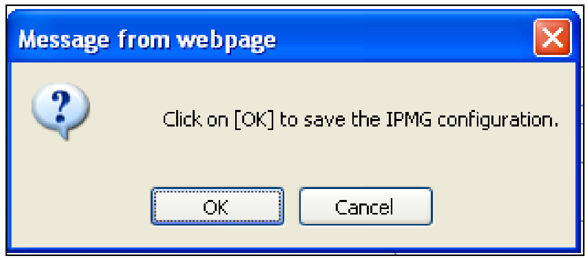


**Step 6** – Scroll down and click on **Codec T.38 FAX**. Note that T.38 is enabled by default.

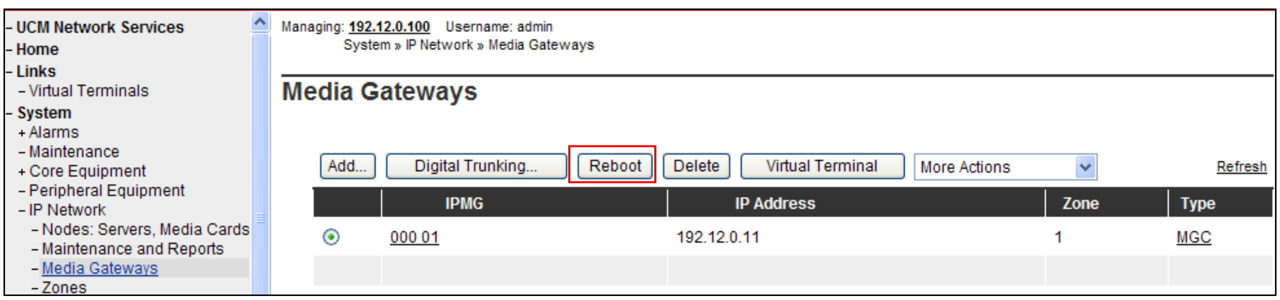



**Step 7** – If changes are made to any of these settings, click on **Save** (not shown).

**Step 8** – A dialog box will open. Click on **Ok**.



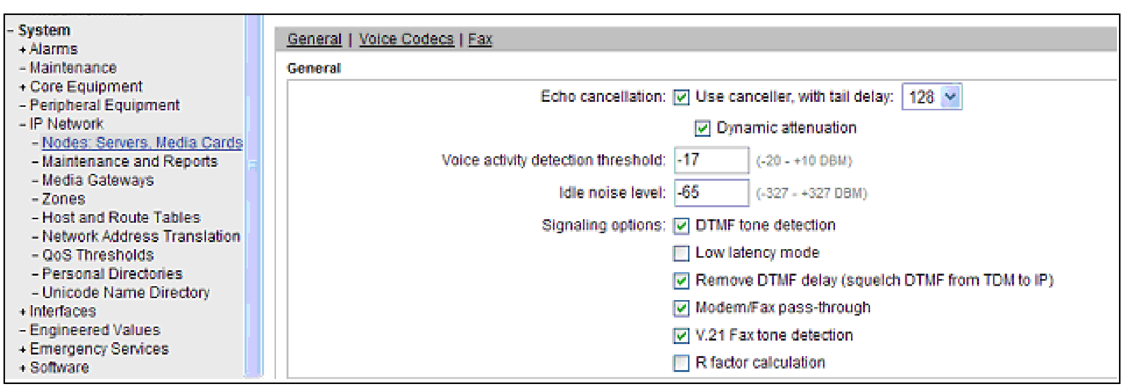

**Step 9** –Select next to the Media Gateway ID (e.g. 000 01), and click on the **Reboot** button. The Media Gateway will reboot and deploy the new configuration.



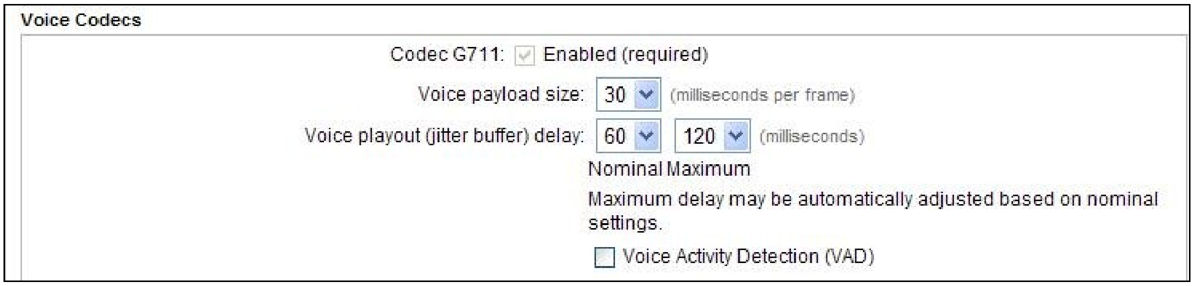

## 5.7.2 IP Telephony Node Codec Configuration

**Step 1** – As shown in **Section 5.1**, **Step 1** expand **System → IP Network**, select **Node, Server, Media Cards**, and select **IP Telephony Node Id "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.

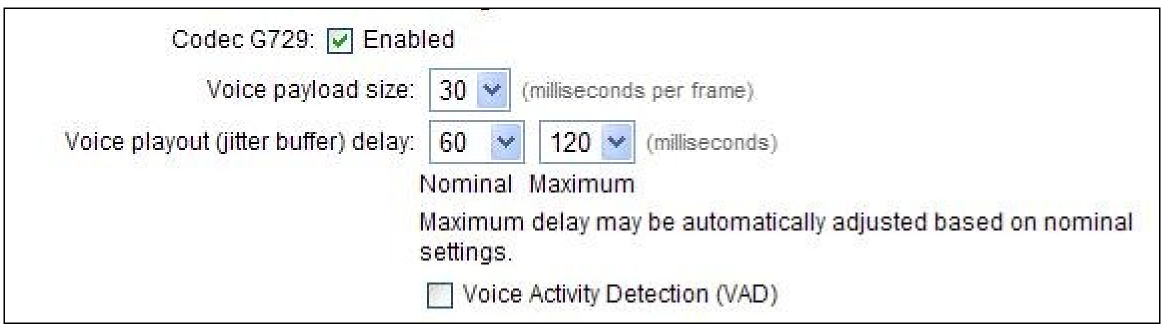


**Step 2** - Use the scroll bar on the right to find the area with heading **Voice Codecs**. Set the **Voice payload size** to **30**. Note that **Codec G.711** is enabled by default.

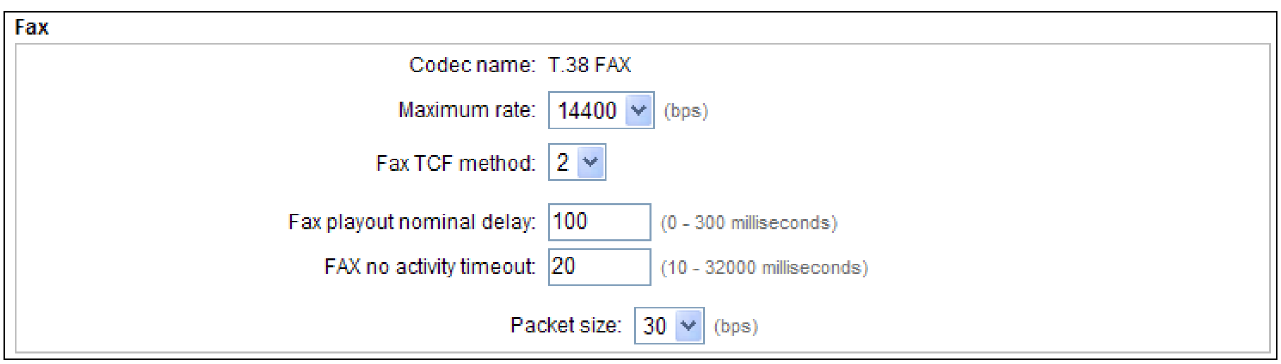



**Step 3** – Scroll down to the G729 codec and check the selection box. Set the **Voice payload size** to **30.**

**Note** – Although not set in the reference configuration, annexB=yes may be enabled by selecting the **VAD** (Voice Activity Detection) box.




**Step 4** - Scrolling further down, note that T.38 fax is enabled by default. Verify the **Maximum Rate** is set to **14400**.




**Step 5** – Click on **Save** and then follow **Steps 8** through **12** in **Section 5.3** to save the configuration.

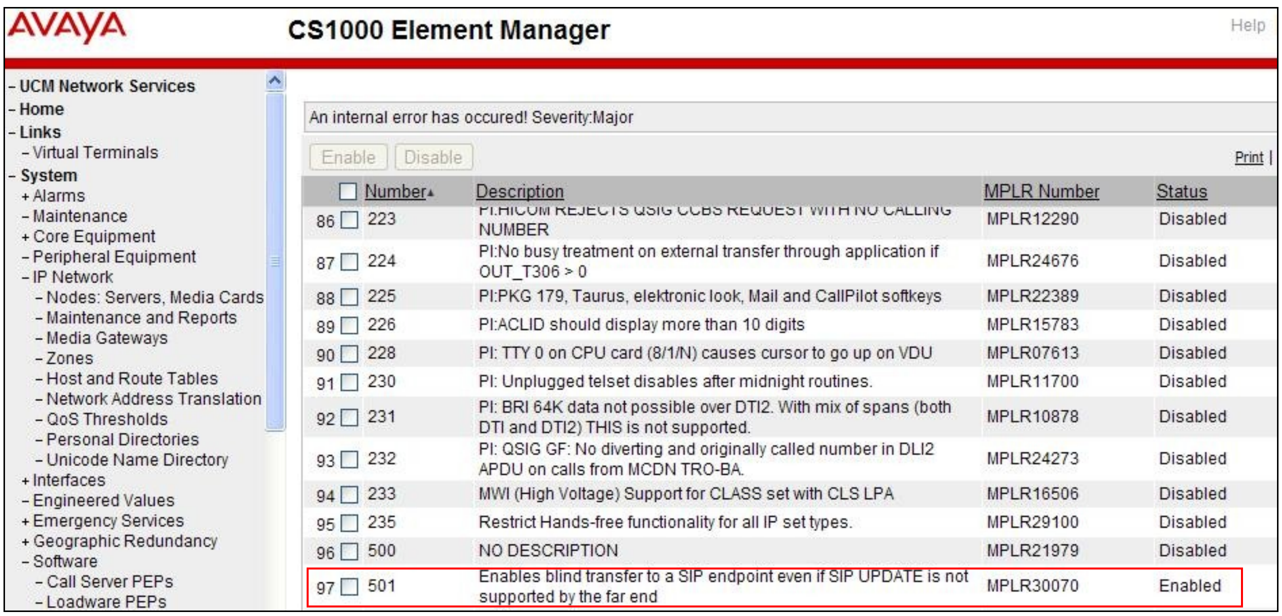
## 5.8. Enabling Plug-Ins for Call Transfer Scenarios

Plug-ins allow specific CS1000E software feature behaviors to be changed. In the testing associated with these Application Notes, plug-in 501 was required for successful completion of Unattended Transfer calls (see **Section 2.2.1**).

**Step 1** - To view or enable a plug-in, from the left navigation menu, expand **System → Software**, and select **Plug-ins** (not shown). In the right side screen, a list of available plug-ins will be displayed along with the associated MPLR Number and Status. Use the scroll bar on the right to scroll down so that Plug-in "**501**" is displayed as shown in the screen below.

**Step 2** - If the **Status** is "Disabled", select the check-box next to Number 501 and click the **Enable** button.

**Note** - Enabling plug-in 501 will allow the user to complete the transfer while the call is in a ringing state, but no audible ring back tone will be heard after the transfer is completed. Without plug-n 501 enabled, Unattended Transfer with not work.



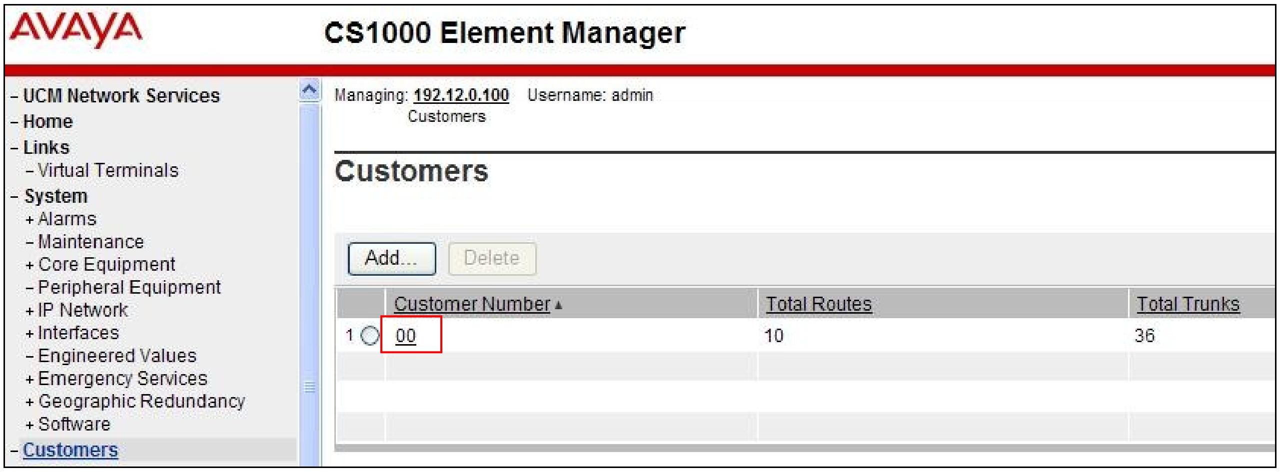

## 5.9. Customer Information

In the reference configuration, specific calling number information is required based on the destination of the call. For Calls to the AT&T IP Flexible Reach service, AT&T assigned DIDs are required.

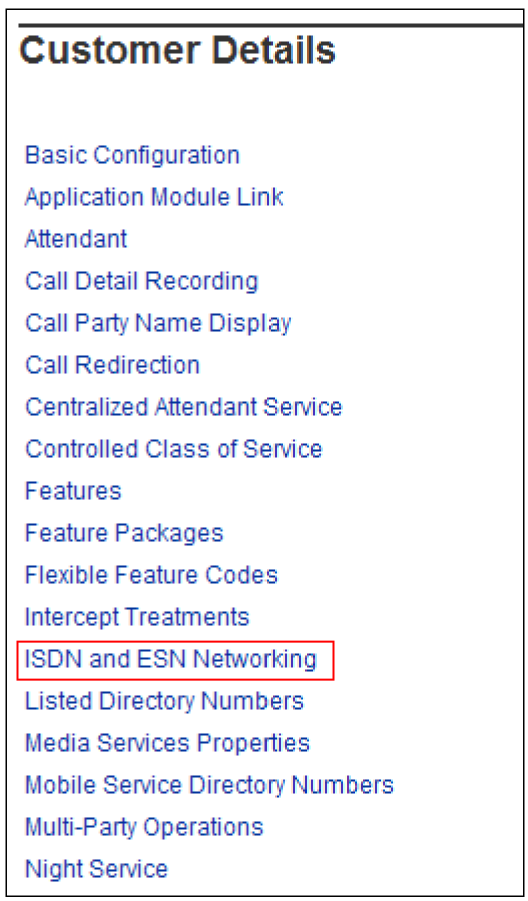
### 5.9.1 Calling Number Provisioning for call to the AT&T IP Flexible Reach Service

The AT&T IP Flexible Reach service expects to see service assigned DID (Direct Inward Dialing) numbers in the SIP origination headers (e.g. From and PAI). In the reference configuration these were 10 digit numbers associated with the local NPA (Note – For security, sample numbers are shown in this document).

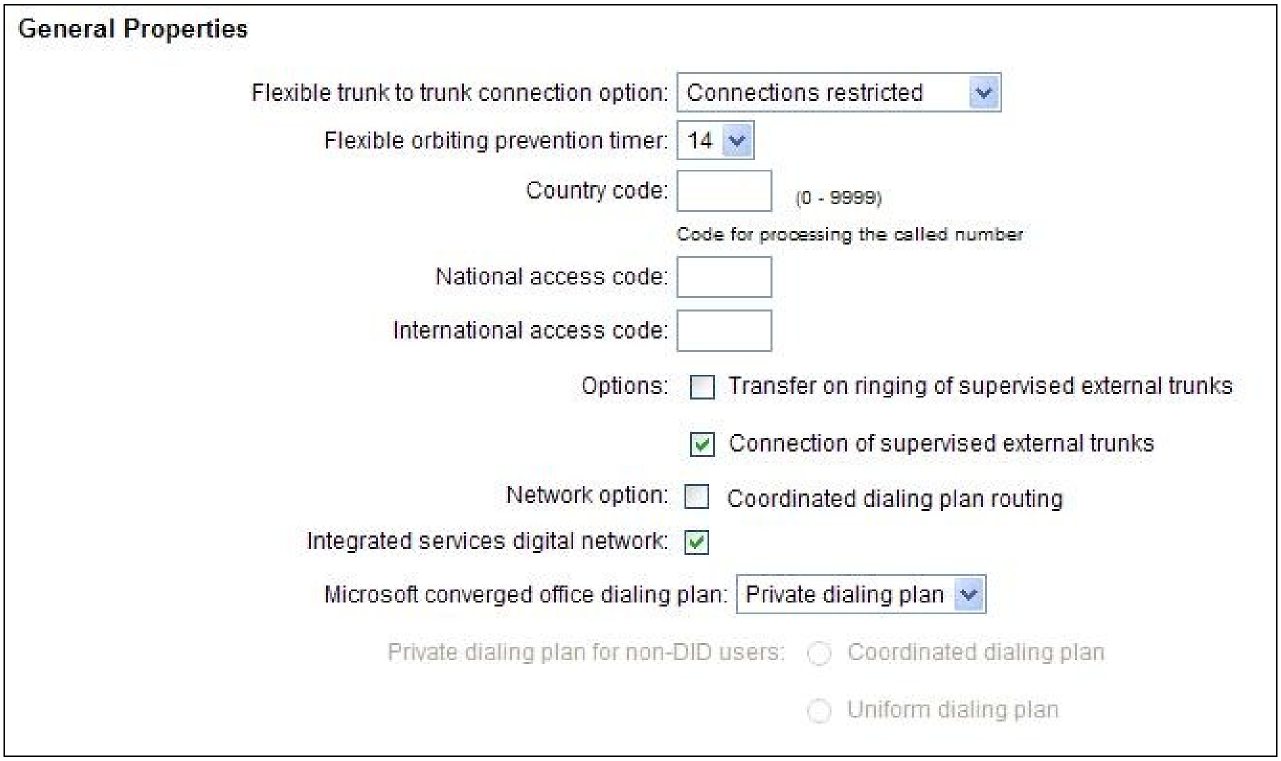
**Step 1** - Select **Customers** from the left navigation menu, click on the appropriate **Customer Number** (e.g. **00)**




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

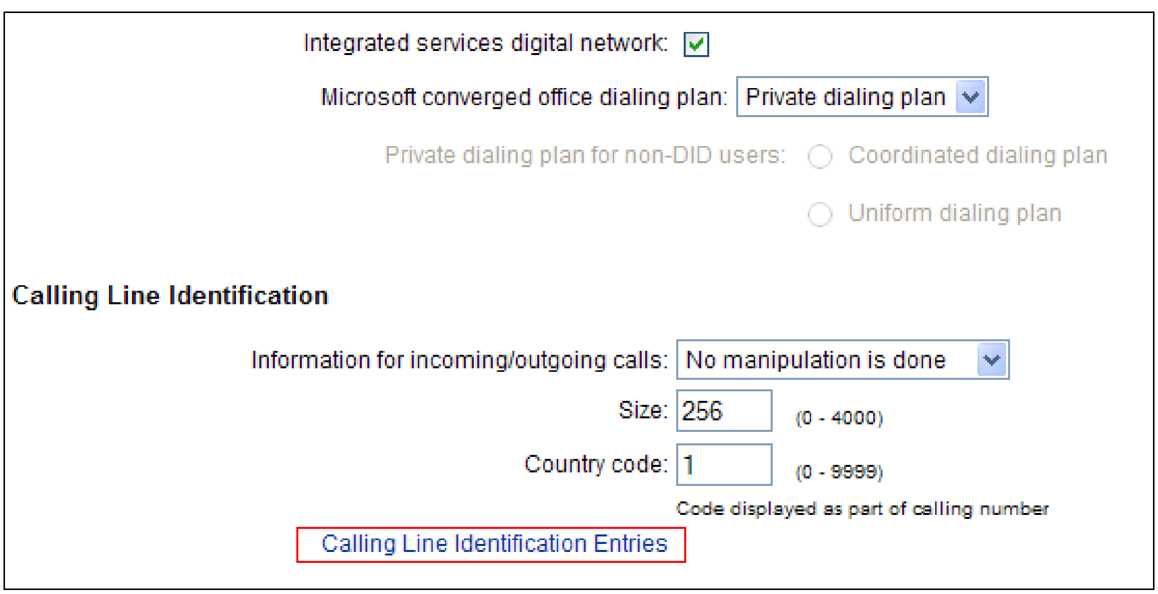


The ISDN and ESN Networking screen will open. As a reference, the following screen shows the **General Properties** used in the reference configuration.



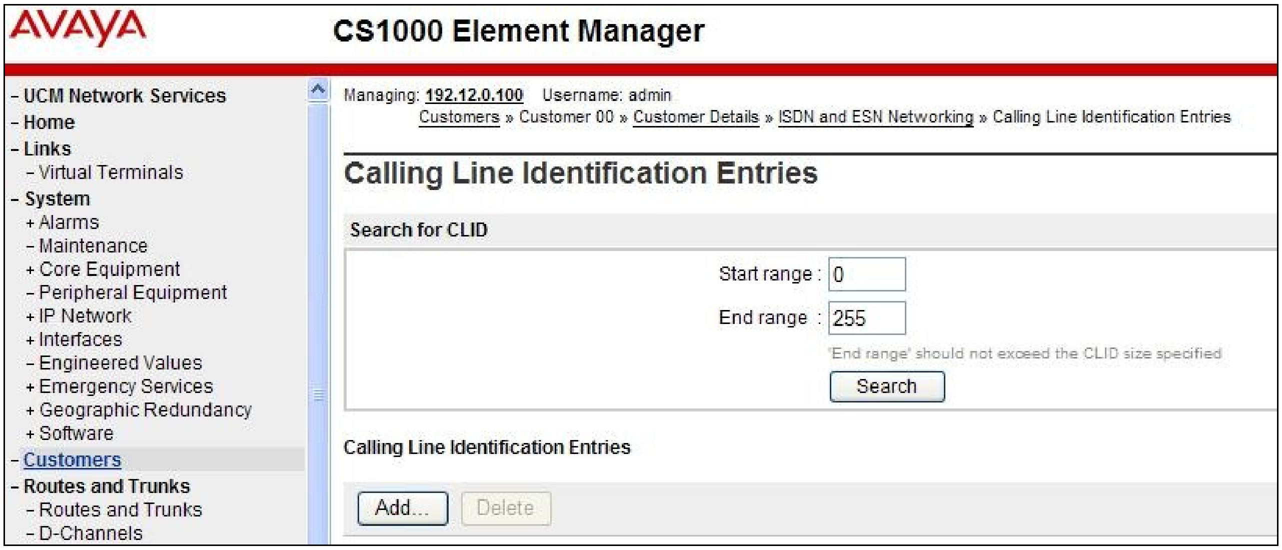

**Step 3** - Scroll down from **General Properties** to the **Calling Line Identification** section and note the value in the **Size** parameter (e.g. **256**).

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

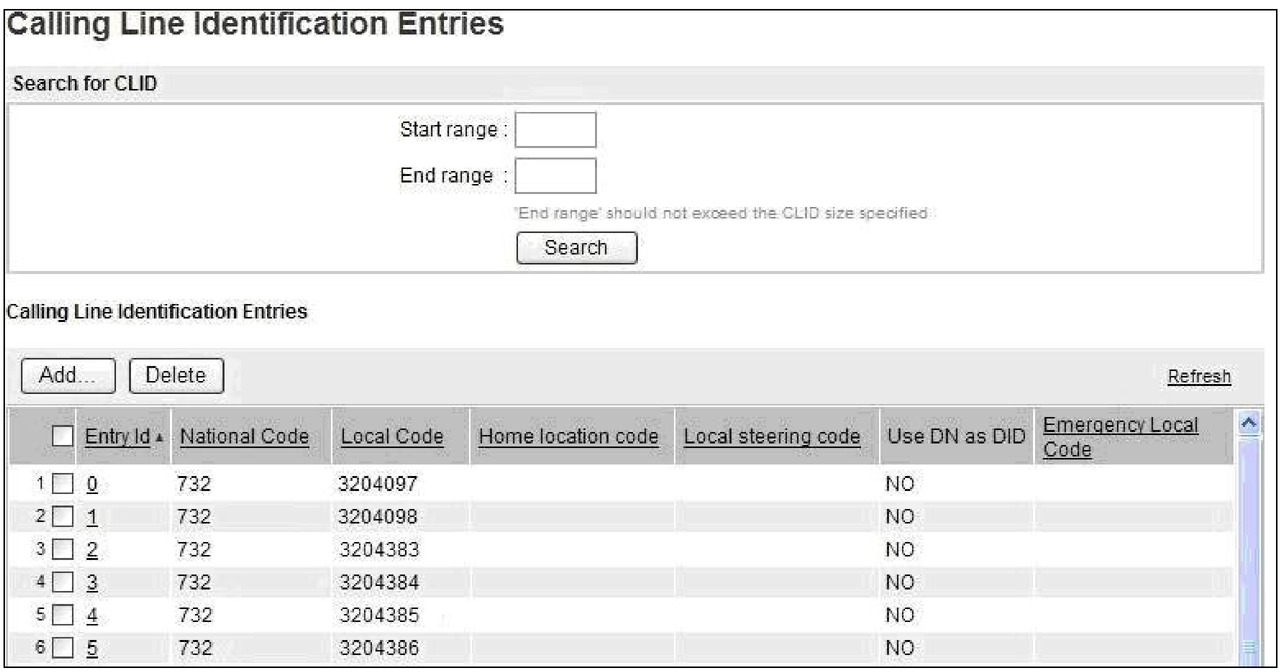


The **Calling Line Identification Entries** page will open.

**Step 5** – In the **Search for CLID** section, enter "**0**" (zero) 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**.

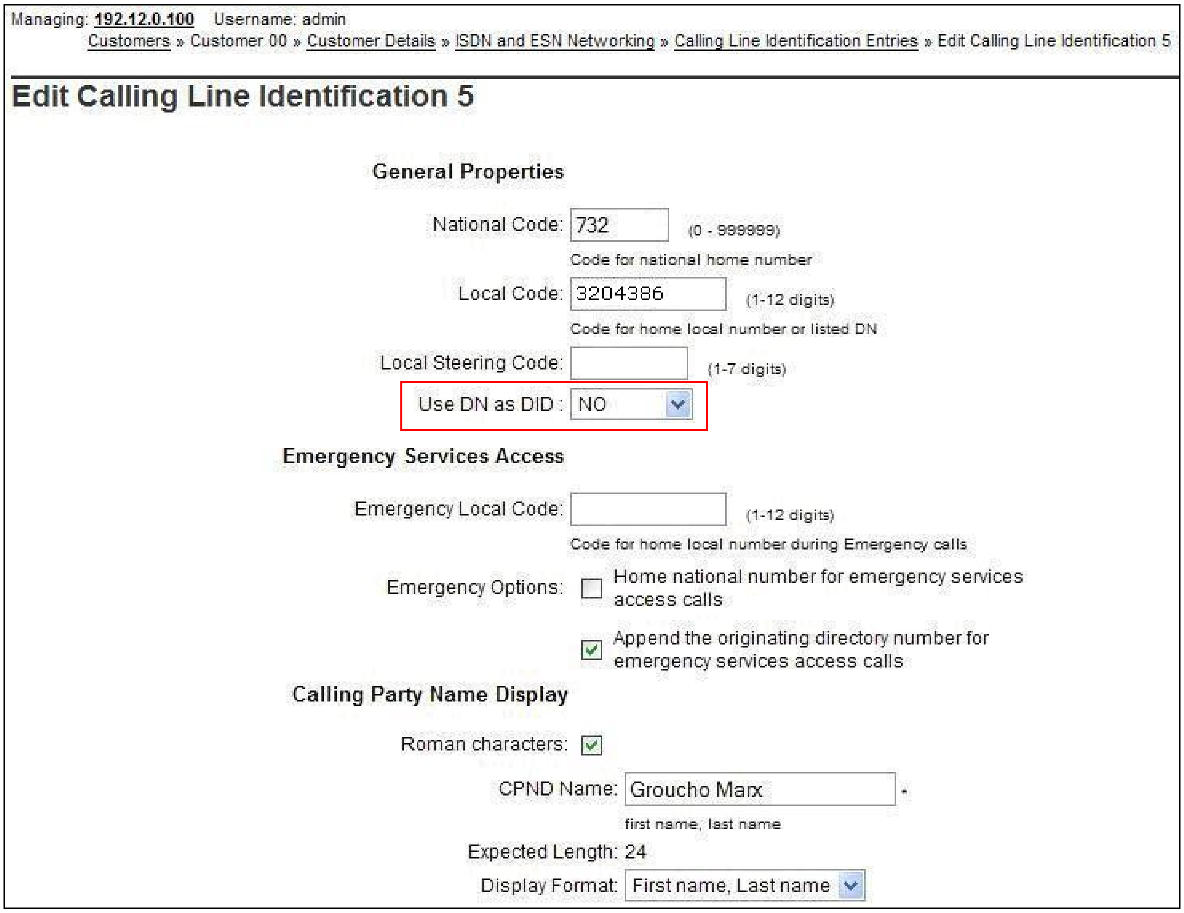



This will display all defined Call Ids. For example CLID 0 will use 732-320-4097




Click on any Entry ID to view or change further details (e.g. **Entry ID 5**).

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). See **Section 5.9.1.2.1.**

### 5.9.1.1 Summary

In summary, to have CS1000E insert the AT&T DID in the origination headers for calls to AT&T via the SIP trunk to Session Manager:

- Customers = 00 (**Section 5.8.1**)
  - ISDN & ESN Networking
    - Calling Line Identification Entries
      - CLID Search
        - Start = 1
        - End = 255
          - Entry ID 5
            - National code = 732
            - Local code - 3204386
            - Use DN as DID = NO
- Phones (**Section 5.9.1**)
  - DN 4094 (select TN)
    - Key 0
      - CLID = 5

## 5.10. CS1000E Stations

This section is not intended to be prescriptive, but simply illustrates a sampling of a telephone station defined in the sample configuration.

### 5.10.1 Example IP UNIStim Phone DN 4094,

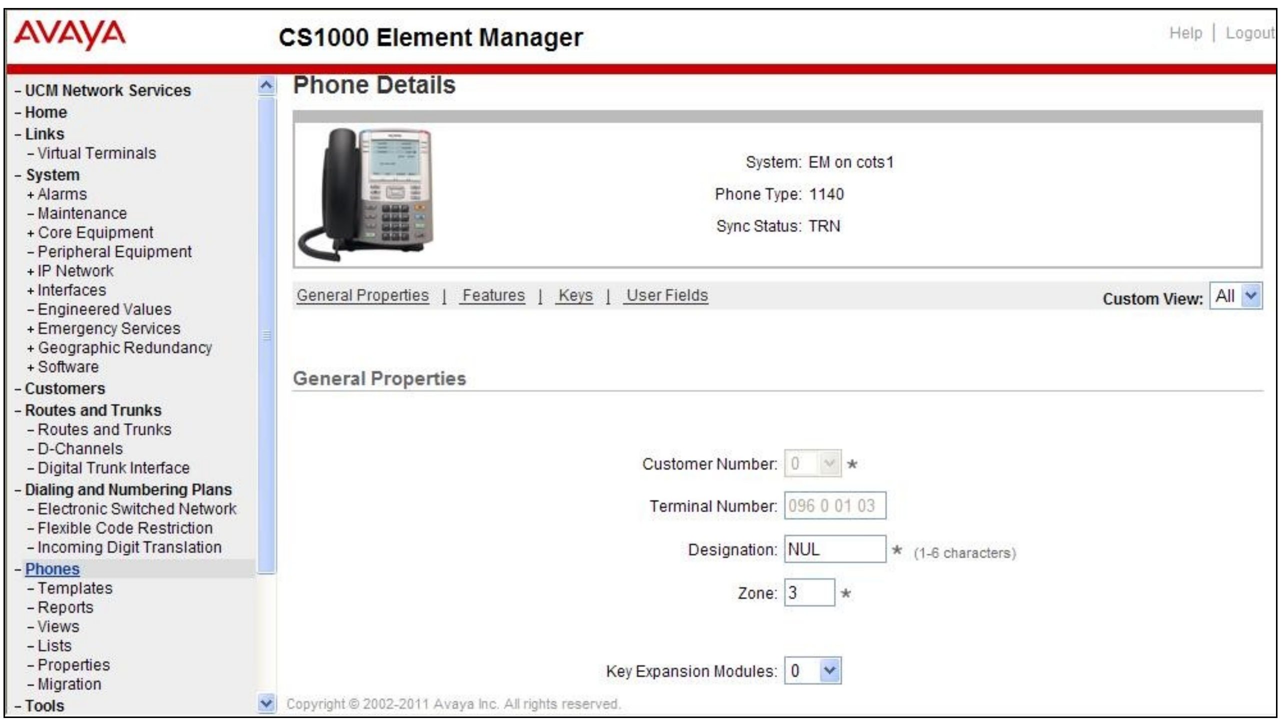
The following screen shows basic information for an IP UNIStim phone in the reference configuration.

**Step 1** – Select **Phones** from the menu. The **Search For Phones** screen will open.

**Step 2** - **Select Criteria** = **Prime DN** and enter a DN in the value field (e.g. **4094**). Click on **Search**.

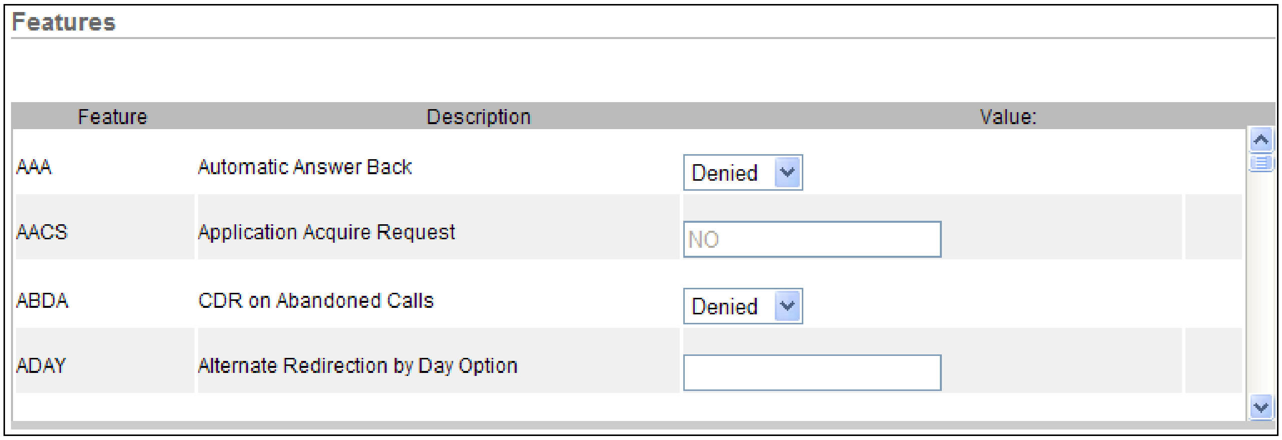
**Step 3** – Click on the TN value (e.g. **096 0 01 03**). The **Phone Details** form will open. Note that the telephone type is an 1140 and that it is defined in Zone 3. A call between this telephone and another telephone in Zone 3 will use a "best quality" strategy (see **Section 5.5**) and therefore can use G.711MU. If this same telephone calls out to the PSTN via the SIP trunk, the call would use a "best bandwidth" strategy, and the call would use G.729A.

**Note** – SIP trunk calls to the AT&T IP Flexible Reach network will use Zone 5, as defined in **Section 5.6**.




#### 5.10.1.1 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. All of the features described below are found by scrolling through this section.

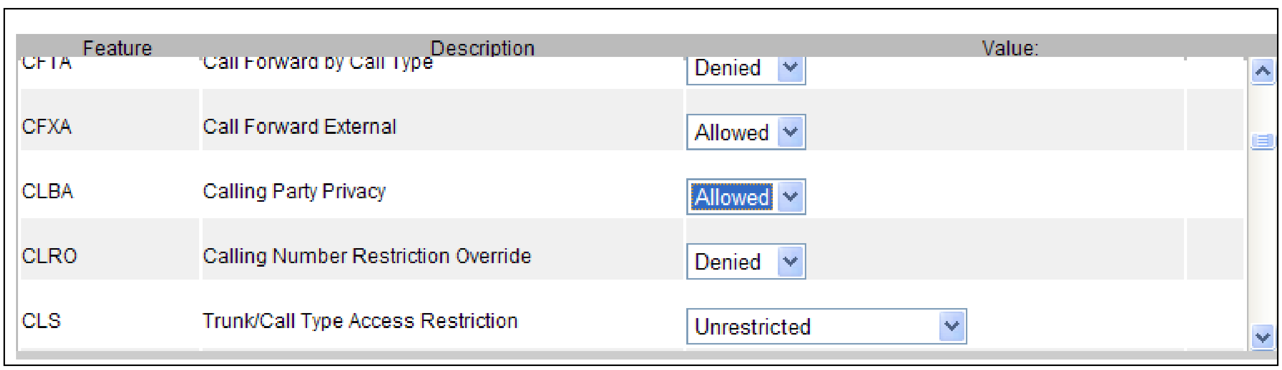


**Features**

| Feature | Description | Value: |
|---|---|---|
| AAA | Automatic Answer Back | Denied |
| AACS | Application Acquire Request | NO |
| ABDA | CDR on Abandoned Calls | Denied |
| ADAY | Alternate Redirection by Day Option | |

### 5.10.1.1.1 Requesting Privacy

One means to have a CS1000E station request privacy (e.g. Privacy: id header in SIP INVITE) for an outbound call, is to set **CLBA Calling Party Privacy** to "**Allowed**" via the Phone **Features** in Element Manager as shown below.



| Feature | Description | Value: |
|---|---|---|
| CFTA | Call Forward by Call Type | Denied |
| CFXA | Call Forward External | Allowed |
| CLBA | Calling Party Privacy | Allowed |
| CLRO | Calling Number Restriction Override | Denied |
| CLS | Trunk/Call Type Access Restriction | Unrestricted |

**Note** - Another means to have the CS1000E request privacy (i.e., Privacy: id in SIP INVITE) for an outbound call is to set **DDGA Present/Restrict Calling Number** to "Denied" (not shown).

### 5.10.1.1.2 Call coverage to Call Pilot

**Step 1** – Set the FDN (Flexible Call Forward No Ans DN) feature to the Call Pilot access extension (e.g. **2080**).

**Step 2** – Set the **FNA** (Call Forward No Answer) feature to **Allowed**.

**Step 3** – Set the **Hunt** (Hunt DN - All Calls, or Internal Calls for CFTA) feature to the Call Pilot access extension (e.g. **2080**).

**Note** - The phone Key **MWK** (Message Waiting) is also required (see **Section 5.9.1.2.3** below).

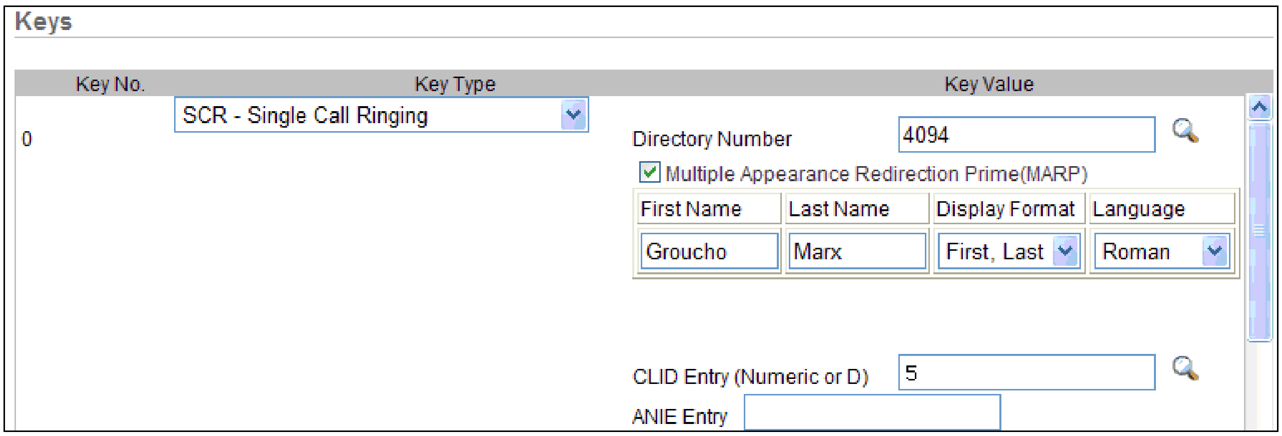
#### 5.10.1.2 Keys

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

##### 5.10.1.2.1 Key 0 - Single Call Appearance

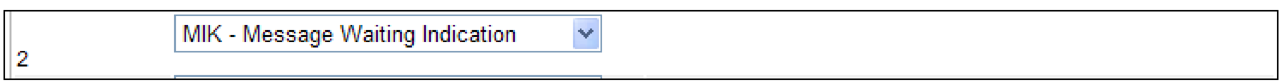
This key defines the first call appearance on the telephone.

**Note** – The **CLID Entry (Numeric or D)** field is where the CLID defined in **Section 5.8** is associated with this station. In the reference configuration telephone station 4094 was assigned CLID 5 and therefore will use 7323204386 as its calling number.



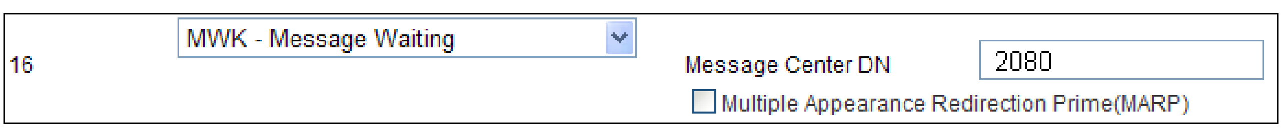

##### 5.10.1.2.2 Key 2 – Message Waiting Indicator

This defines the MWI lamp.



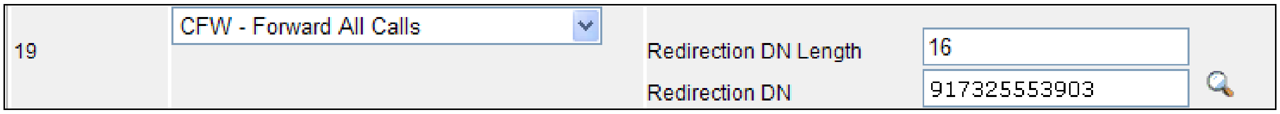

##### 5.10.1.2.3 Key 16 - Message Waiting

This key defines the extension CS1000E will dial to reach the messaging system.



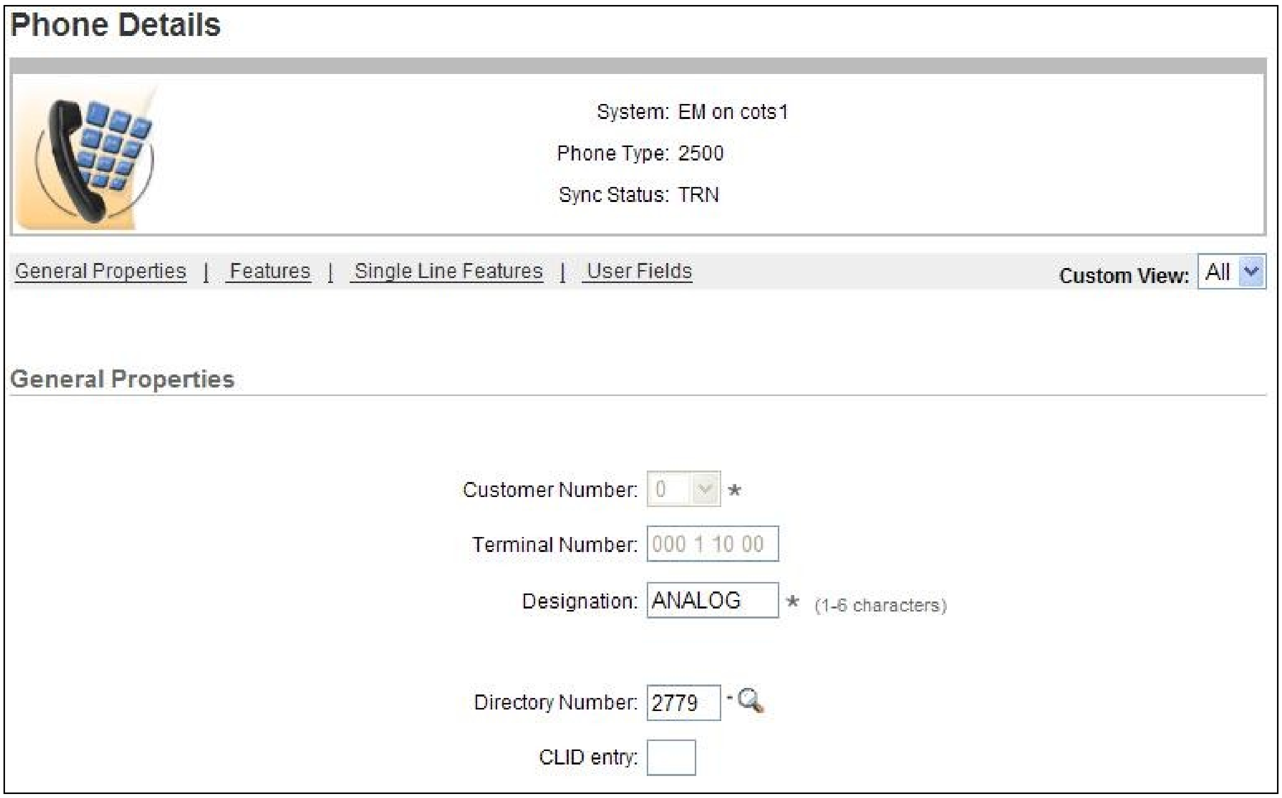

##### 5.10.1.2.4 Key 19 - Forward All Calls

This key defines an alternate destination to redirect inbound calls to this station.




### 5.10.2 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.




## 5.11. Changing RFC2833 DTMF Telephone Event Type

The CS1000E uses RFC2833 DTMF Telephone Event type 101 by default. The AT&T IP Flexible Reach service uses 100. While having asymmetric telephone event types is permitted, this may cause issues in some call scenarios. If an issue occurs, the CS1000E value may be changed to 100 as follows:

**Step 1** – From a CS1000E console connection (e.g. serial interface), press the ctrl key and enter "**pdt**". The system will return:

```
PDT login on /tyCo/0
```

```
Username:
```

**Step 2** – Enter the appropriate login. 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.

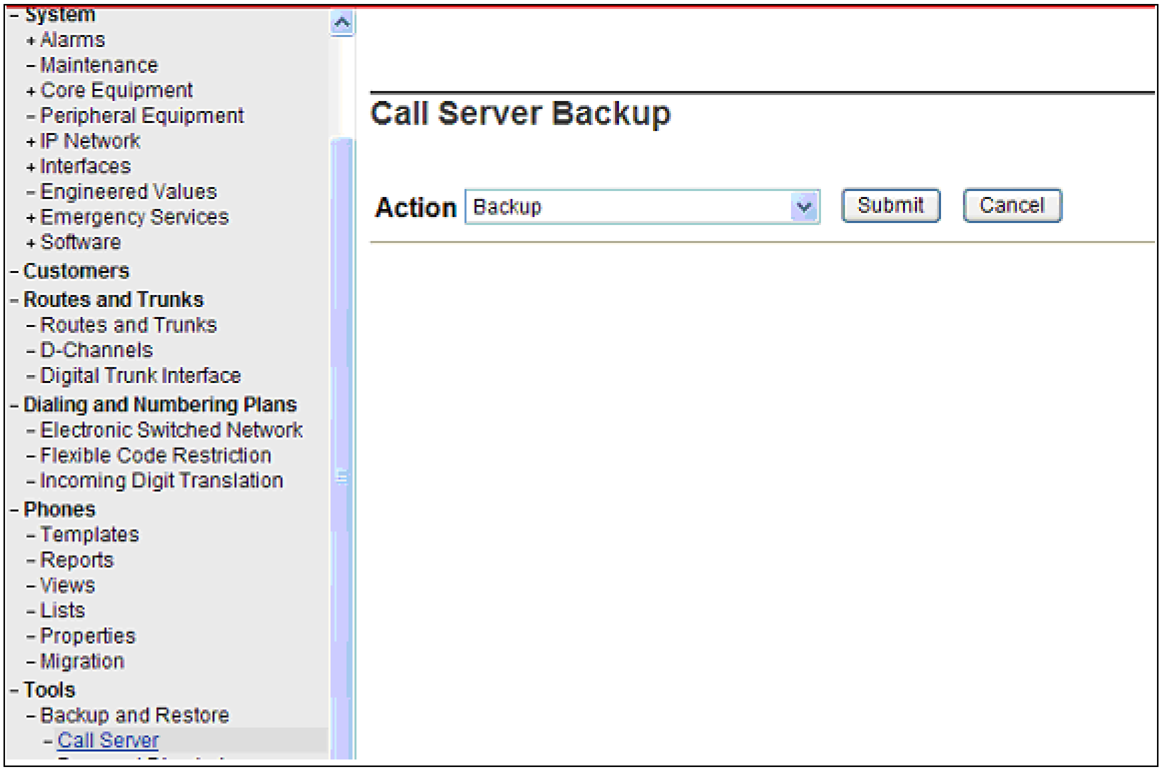
```
pdt>
```

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

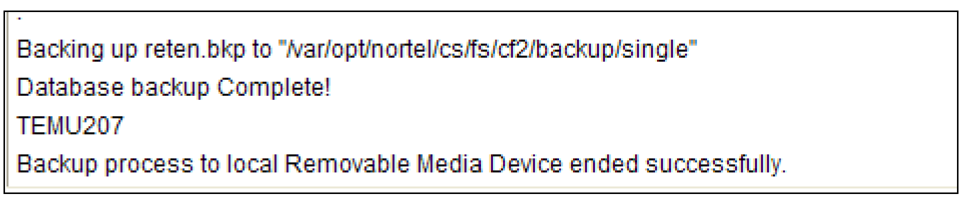
| |
|---|
| **NOTE** – If the CS1000E is rebooted, this command will be cleared and the system will use telephone event 101 again. This command must be re-entered. |

## 5.12. 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.




The configuration of Avaya Communication Server 1000E is complete.

# 6. Configure Avaya Aura® Session Manager Release 6.1

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 12**.

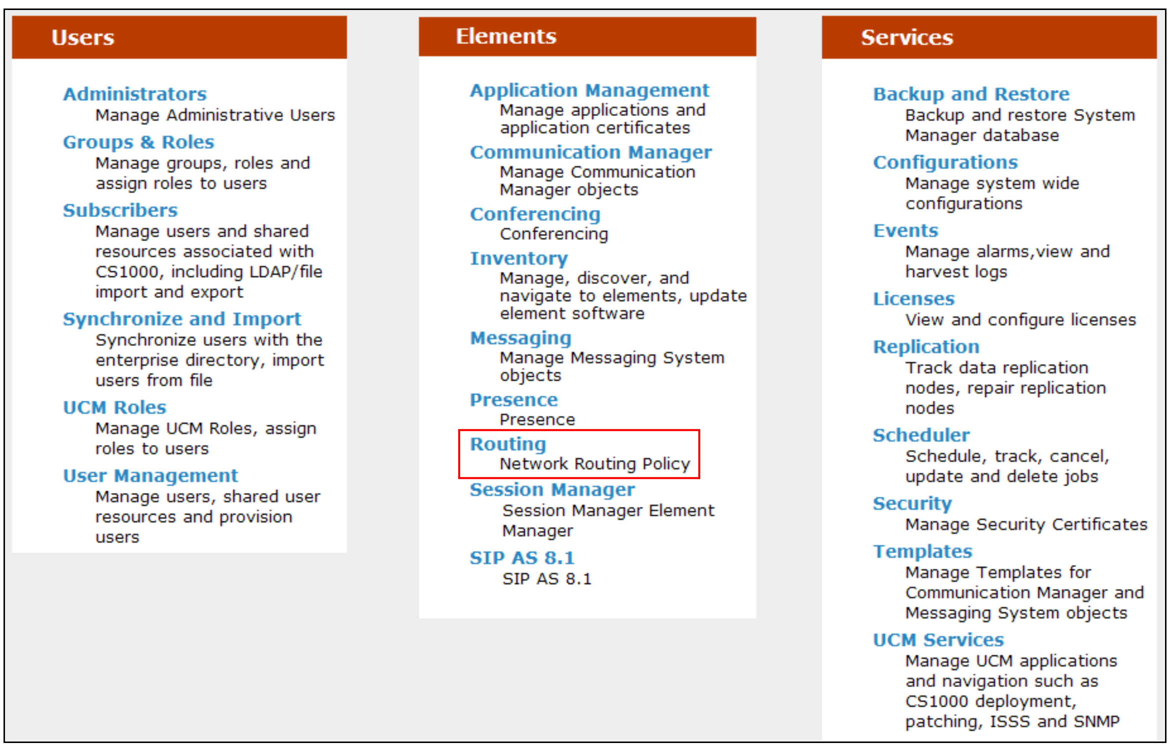
This section provides the procedures for configuring Session Manager to receive calls from and route calls to the SIP trunk between Avaya Communication Server 1000E and Session Manager, and the SIP trunk between Session Manager and the Avaya Aura® SBC.

The following administration activities will be described:

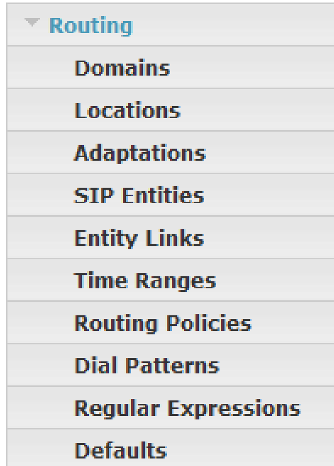
- Define SIP Domain
- Define Locations for Avaya Communication Server 1000E and for the SBC
- Configure the Adaptation Modules that will be associated with the SIP Entities for Avaya Communication Server 1000E and the SBC
- Define SIP Entities corresponding to Avaya Communication Server 1000E and the SBC
- Define Entity Links describing the SIP trunk between Avaya Communication Server 1000E and Session Manager, and the SIP Trunk between Session Manager and the SBC.
- Define Routing Policies associated with the Avaya Communication Server 1000E and the SBC.
- Define Dial Patterns, which govern which routing policy will be selected for call routing.

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

In the **Log On** screen (not shown), enter appropriate **User ID** and **Password** and press the **Log On** button. Once logged in, a Release 6.1 **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.

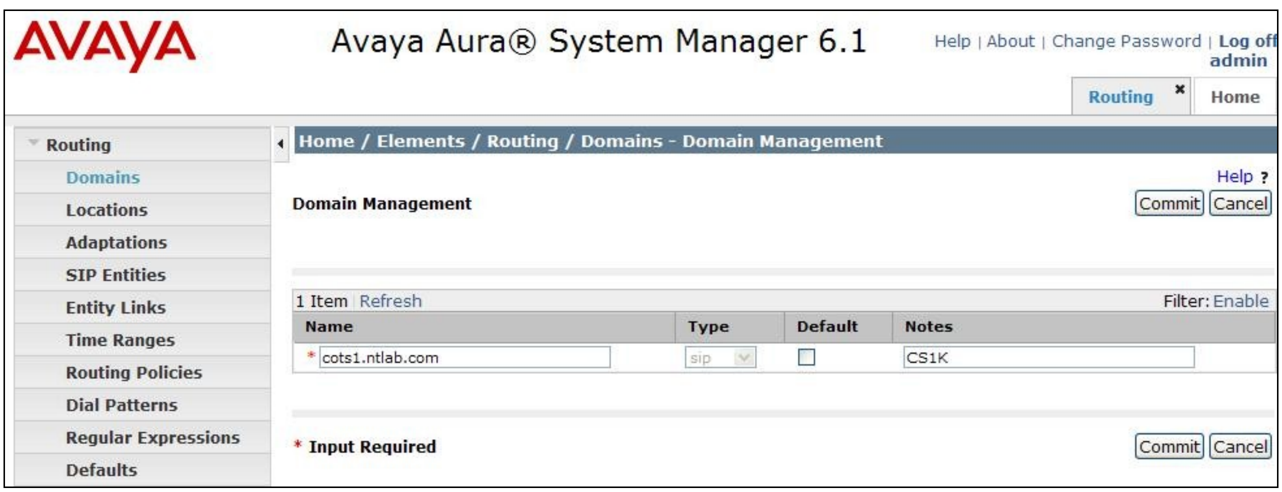



## 6.1. SIP Domain

**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 and use default values for remaining fields**.**

- **Name** Enter the enterprise SIP Domain Name. In the sample screen below, "**cots1.ntlab.com**" is shown.
- **Type** Verify "**SIP**" is selected.
- **Notes** Add a brief description. [Optional]




**Step 3** - Click **Commit** to save.

**Note** - 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. 192.168.10.x for all devices on a particular subnet), or individual devices (e.g. 192.168.10.10 for a devices' IP address). In the reference configuration, the CS1000E, and the Avaya Aura® SBC were each defined as individual Locations.

### 6.2.1 Location for Avaya Communication Server 1000E

**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**.**

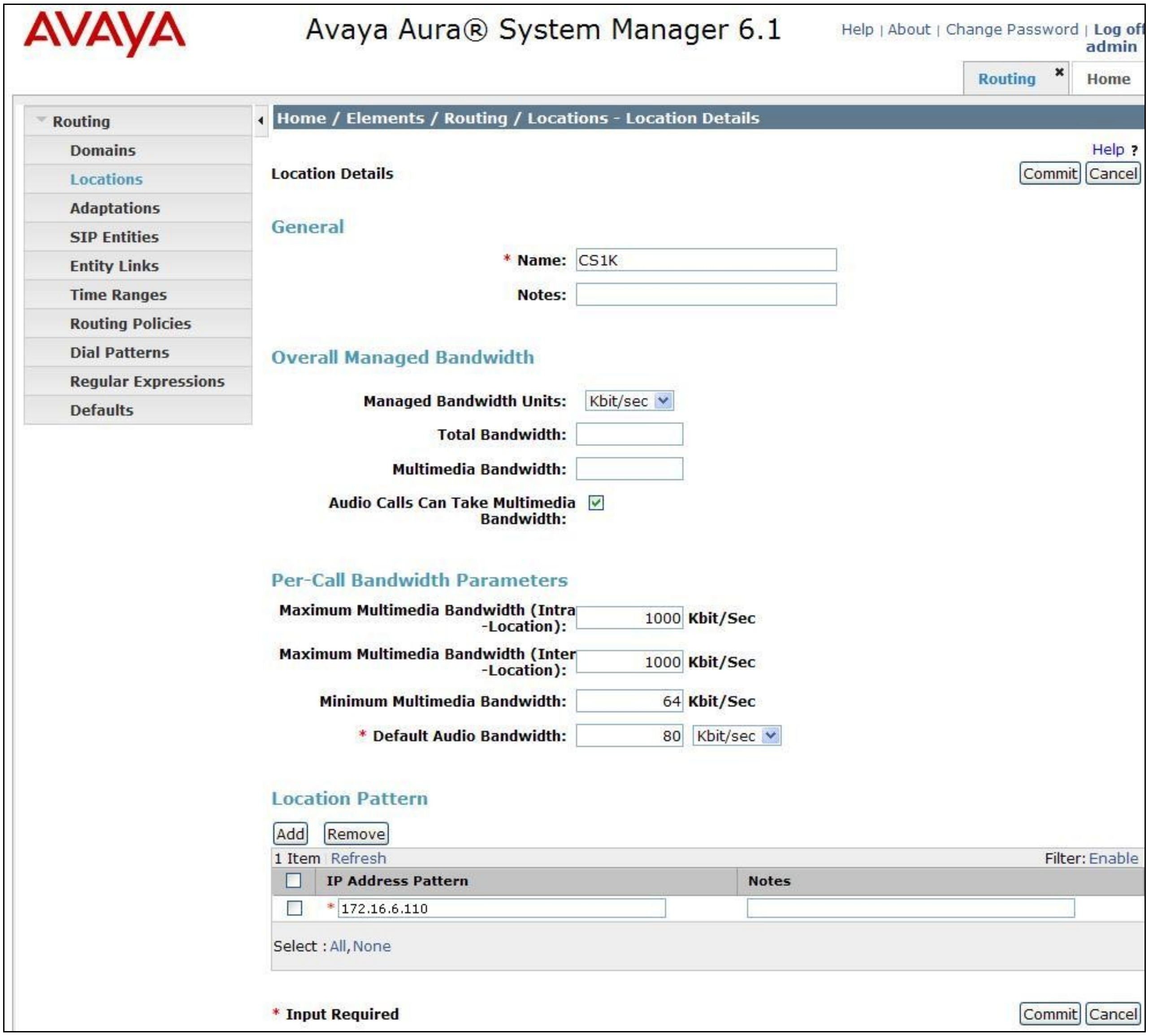
- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description. [Optional]

**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. **172.16.6.110**).
- **Notes** Add a brief description. [Optional]

**Step 3** - Click **Commit** to save.

The screen below shows the top portion of the screen for the Location defined for Avaya Communication Server 1000E.




## 6.2.2 Location for the Avaya Aura® Session Border Controller

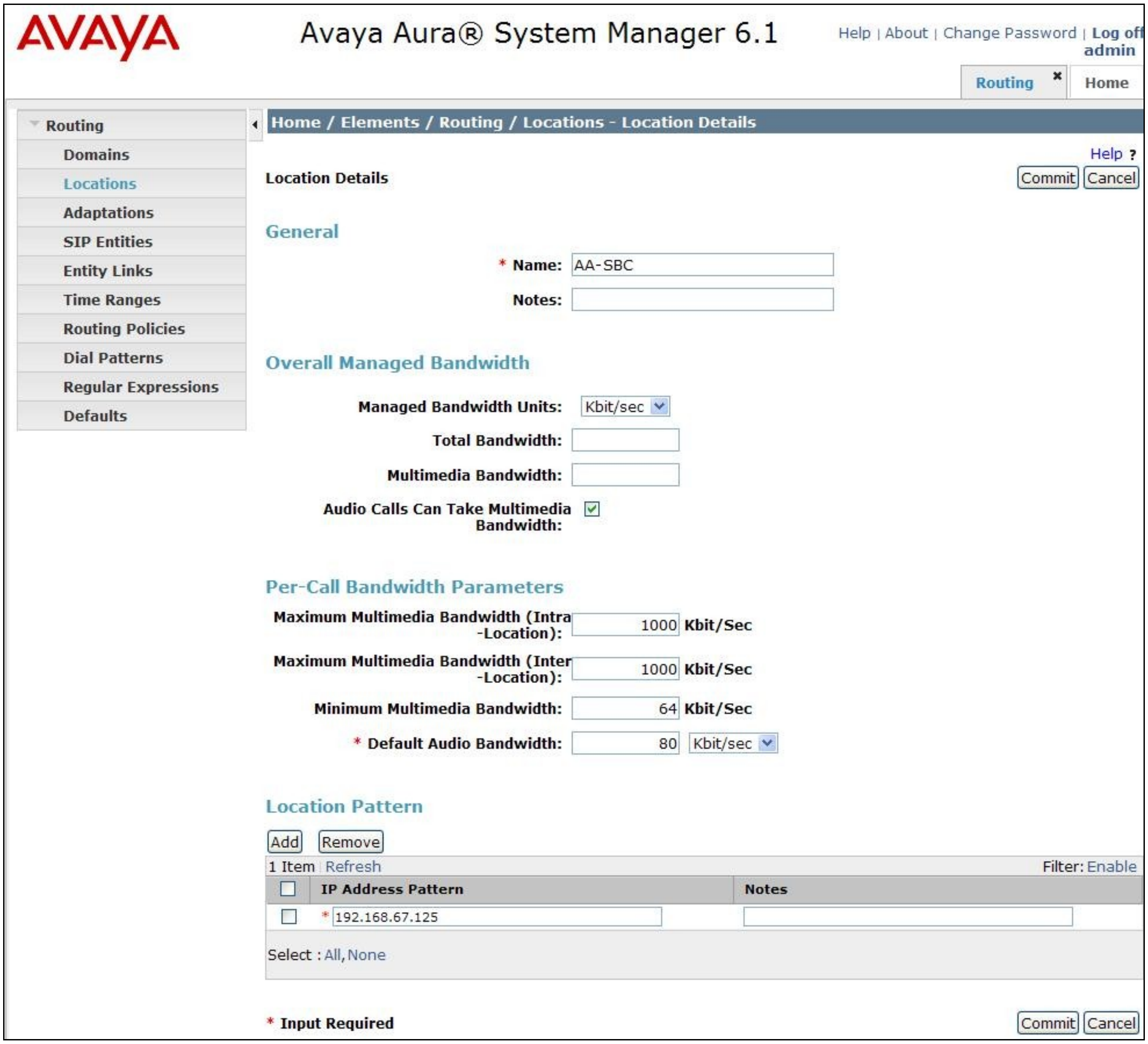
**Step 1** - Select **Locations** from the left navigational menu. Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

- **Name:** Enter a descriptive name for the location.
- **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 Avaya Aura® SBC location (e.g. **192.168.67.125**).
- **Notes** Add a brief description. [Optional]

**Step 3** - Click **Commit** to save.




## 6.3. Configure Adaptations

Session Manager can be configured to use an Adaptation Module designed for Avaya Communication Server 1000E to convert SIP headers in messages sent by Avaya Communication Server to the format used by other Avaya products and endpoints. In the reference configuration the following adaptations was used.

- **DiversionTypeAdapter** – This adaptation is used to convert History-Info headers sent by the CS1000E in certain outbound calls to AT&T (which are not supported by the AT&T IP Flexible Reach service), to Diversion Headers. This is required for call scenarios such as Call Forwarding.
- **CS1000Adapter** – This adaptation is used to provide translation between CS1000E generated History-Info headers into formats used by other Avaya products and endpoints.

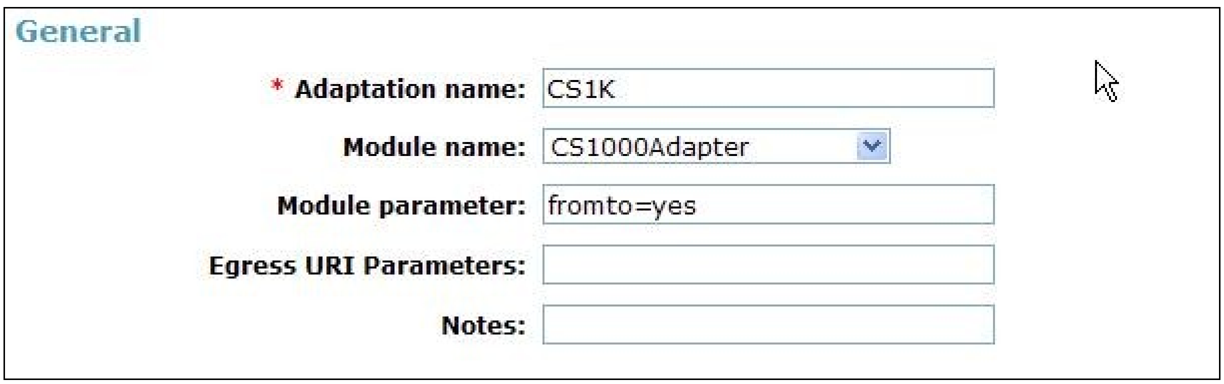
- **DigitConversionAdapter** – This adaptation is used to modify digit strings in the Request-URI. Note that the adaptation functionality is included in all other adaptations.

In addition, Module parameters **odstd** (to modify destination domain or IP addressing), **osrcd** (to modify source domain or IP addressing, **MIME=no** (to remove unnecessary CS1000K SIP headers), and **fromto=yes** (to modify the From and To header) are specified.

### 6.3.1 Adaptation for Avaya Communication Server 1000E Entity

**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., "CS1000")
- **Module Name:** Select "**CS1000Adapter**" from drop-down menu (or add an adapter with name "CS1000Adapter" if not previously defined)
- **Module Parameter:** Enter **fromto=yes** (Note – this parameter is set so that the correct To header information is provided for calls to Call Pilot).




**Step 2** - Scrolling down, in the **Digit Conversion for Incoming Calls To SM** section, click **Add** to configure entries for calls from CS1000E users to AT&T. The text below and the screen example that follows explain how to use Session Manager to convert between CS1000K extensions and AT&T IP Flexible Reach DIDs.

- **Matching Pattern** Enter a CS1000E extension (e.g. 4094).
- **Min** Enter minimum number of digits (e.g. 4)
- **Max** Enter maximum number of digits (e.g. 4)
- **Phone Context** Leave blank.
  **Delete Digits** Enter "**4**", to remove the CS1000E extension digits.
- **Insert Digits** Enter the corresponding AT&T IP Flexible Reach DID(e.g. 7323204383).
- **Address to modify** Select **"both".**

Repeat for any addition CS1000E extensions.

**Digit Conversion for Incoming Calls to SM**

Add | Remove

4 Items | Refresh — Filter: Enable

| ☐ | Matching Pattern | Min | Max | Phone Context | Delete Digits | Insert Digits | Address to modify | Notes |
|---|---|---|---|---|---|---|---|---|
| ☐ | * 4093 | * 4 | * 4 | | * 4 | 7323204385 | both | |
| ☐ | * 4094 | * 4 | * 4 | | * 4 | 7323204383 | both | |
| ☐ | * 4095 | * 4 | * 4 | | * 4 | 7323204385 | both | |

Select : All, None

**Step 3** - Scrolling down, in the **Digit Conversion for Outgoing Calls from SM** section, click **Add** to configure entries for calls from AT&T to access the integrated Call Pilot messaging system. The text below and the screen example that follows explain how to use Session Manager to convert between inbound AT&T IP Flexible Reach DIDs and CS1000K/Call Pilot extension (2090).

- **Matching Pattern** Enter AT&T IP Flexible Reach DIDs (e.g. **7323204384**).
- **Min** Enter minimum number of digits (e.g. 10)
- **Max** Enter maximum number of digits (e.g. 10)
- **Phone Context** Leave blank.
  **Delete Digits** Enter "**10**", to remove the AT&T DID digits.
- **Insert Digits** Enter the corresponding Call Pilot extension (e.g. **2090**).
- **Address to modify** Select **"both".**

Repeat for any addition AT&T DID access to Call Pilot.

**Digit Conversion for Outgoing Calls from SM**

Add | Remove

Filter: Enable

| ☐ | Matching Pattern | Min | Max | Phone Context | Delete Digits | Insert Digits | Address to modify | Notes |
|---|---|---|---|---|---|---|---|---|
| ☐ | * 3143325386 | * 10 | * 10 | | * 10 | 2090 | both | Call Pilot |
| ☐ | * 4386 | * 4 | * 4 | | * 4 | 2090 | both | Call Pilot |
| ☐ | * 7323204384 | * 10 | * 10 | | * 10 | 2090 | both | Call Pilot |

Select : All, None

### 6.3.2 Adaptation for the Avaya Aura® SBC Entity

The message body of an INVITE message sent from the CS1000E will 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. On the production circuit used for testing, AT&T was able to properly parse the Multipart MIME message body, and outgoing calls from the CS1000E to AT&T could be completed successfully without the configuration in this section. Nevertheless, 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 Aura® SBC can be configured to remove these headers as well (see **Section 7.2.5**). Either method is acceptable.

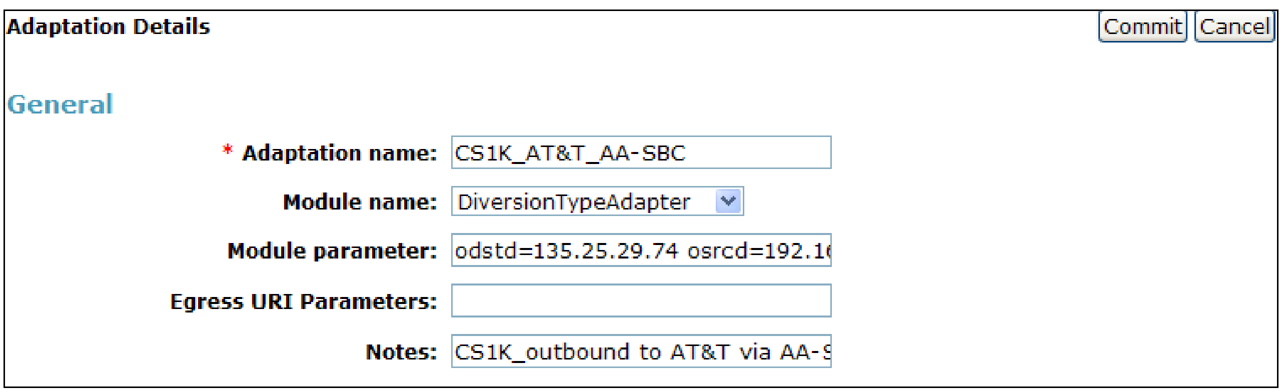
**Step 1** - Select **Adaptations** from the left navigational menu. Click **New (**not shown**).** In the **General** section, enter the following values and use default values for remaining fields**.**

- **Adaptation Name:** Enter an identifier for the Adaptation Module
- **Module Name:** Select "**DiversionTypeAdapter**" from drop-down menu (or add an adapter with name "DiversionTypeAdapter" if not previously defined)
- **Module Parameter:** Enter the following three parameters separated by spaces.
    - Enter "**odstd=**<IP address of the public interface of the Avaya Aura® SBC>" (e.g. **odstd=135.25.29.74**).
    - Enter "**osrcd=**<IP address of the AT&T IP Flexible Reach border element>" (e.g. **osrcd=192.168.64.130**).
    - Enter "**MIME=no**" to remove additional MIME Media Type headers that the CS1000E adds to its SIP signaling.

The entire Module parameter string will appear as:

**odstd=135.25.29.74 osrcd=192.168.64.130 MIME=no**

Note that the entire entry is not visible in the screenshot below.

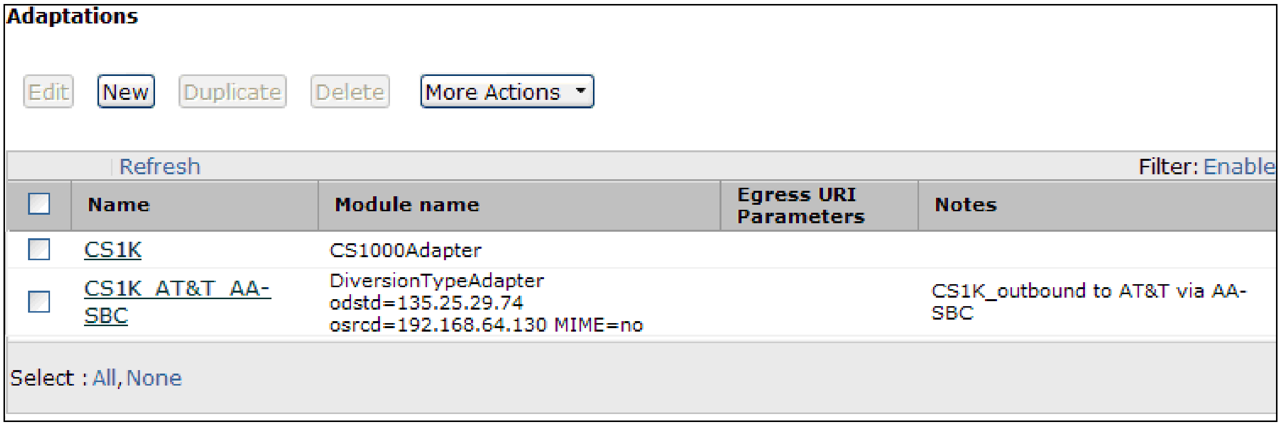



**Note** – Neither **Digit Conversion for Incoming Calls to SM** or **Conversion for Outgoing Calls from SM Digit** were required in the reference configuration for the Avaya Aura® SBC SIP Entity.

**Step 2** - Click **Commit.**

## 6.3.3 List of Adaptations

Select **Adaptations** from the left navigational menu. The completed list of the Adaptation Modules defined for the sample configuration is shown below. In list form, the module parameters assigned to the adapters are more evident than the screens presented in the prior sections.




## 6.4. SIP Entities

SIP Entities must be added for the CS1000E and the Avaya Aura® SBC. 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.

### 6.4.1 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**.**

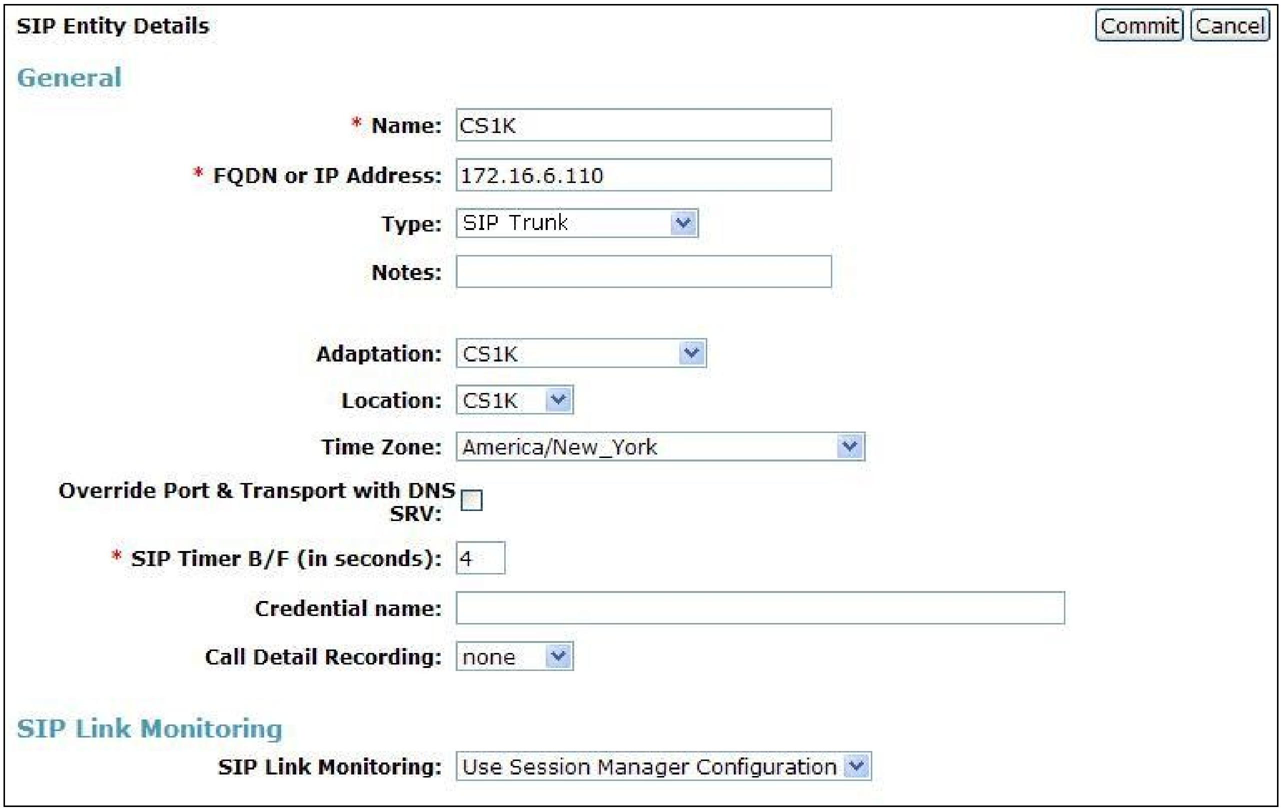
- **Name:** Enter an identifier for the SIP Entity
- **FQDN or IP Address:** Enter the TLAN IP address of the CS1000E Node.
- **Type:** Select "**SIP Trunk**"
- **Notes:** Enter a brief description. [Optional]
- **Adaptation:** Select the Adaptation Module defined in **Section 6.3.1**.
- **Location:** Select the 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** to save the definition of the new SIP Entity.

The following screen shows the SIP Entity defined for theCS1000E in the sample configuration.




### 6.4.2 SIP Entity for the Avaya Aura® SBC

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

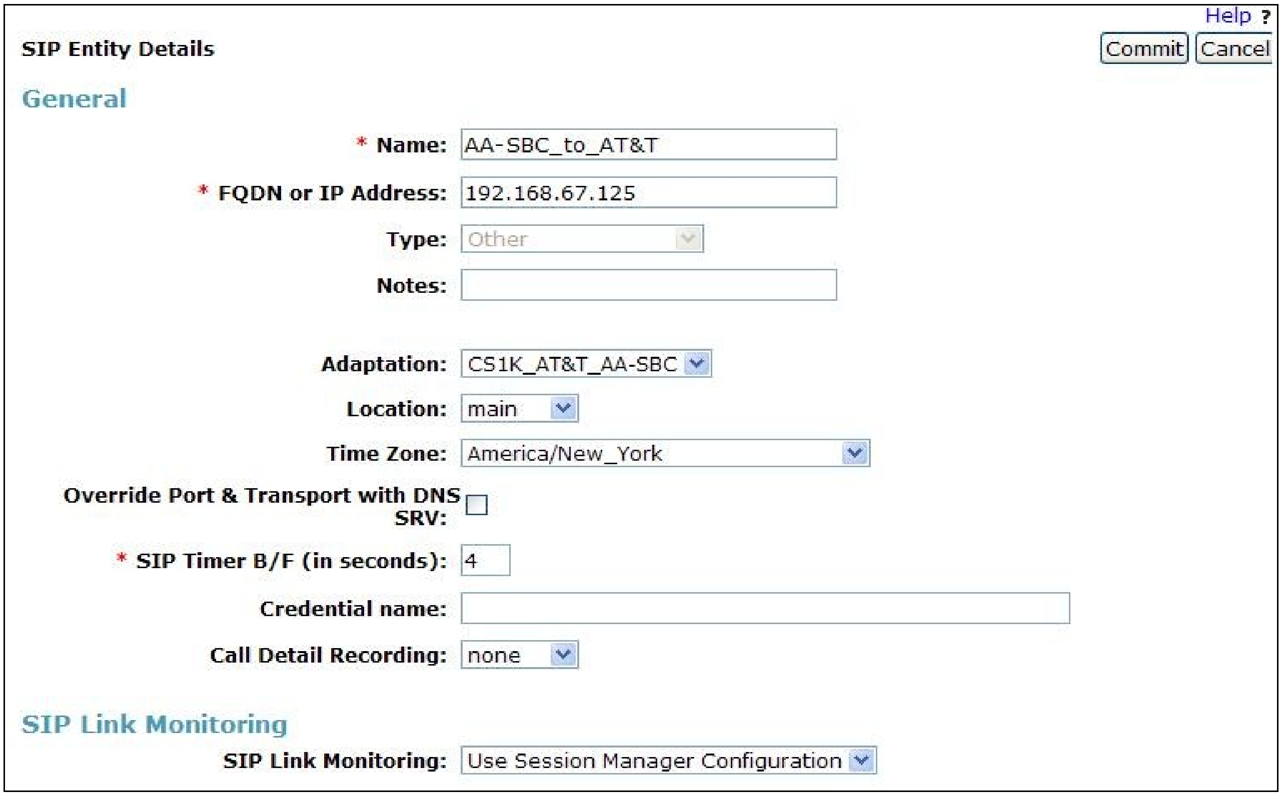
**Step 2** - Click **New (**not shown**).** In the **General** section, enter the following values and use default values for remaining fields**.**

- **Name:** Enter an identifier for the SIP Entity
- **FQDN or IP Address:** Enter the private side IP Address of the SBC.
- **Type:** Select "**Other**"
- **Notes:** Enter a brief description. [Optional]
- **Adaptation:** Select the Adaptation Module defined in **Section 6.3.2**.
- **Location:** Select the Location 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).

The following screen shows the SIP Entity defined for the SBC in the sample configuration.




## 6.5. Entity Links

The SIP trunk between Session Manager and the CS1000E is described by an Entity Link, as is the SIP trunk between Session Manager and the SBC.

### 6.5.1 Entity Link to Avaya Communication Server 1000E Entity

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

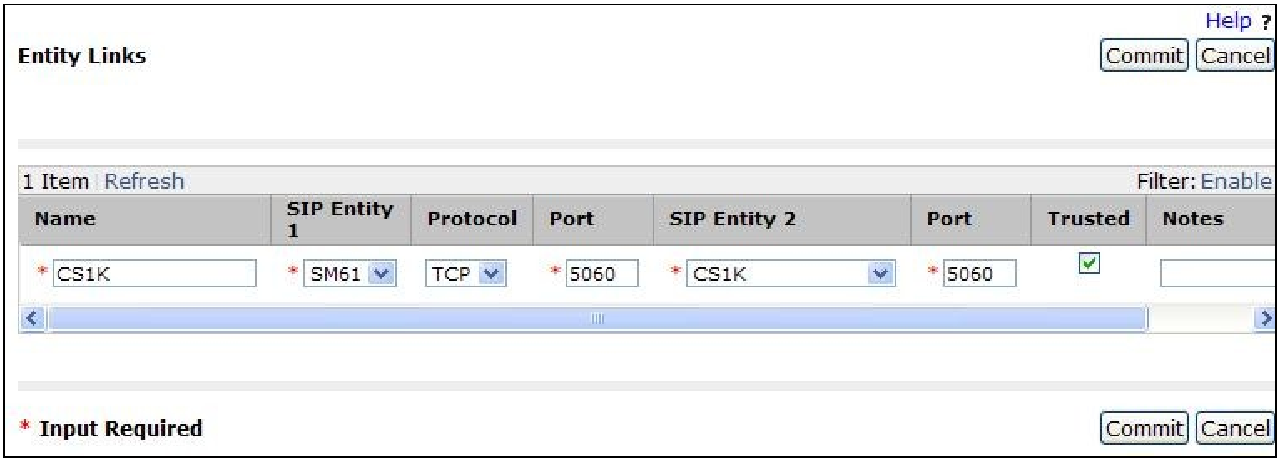
**Step 2** - Click **New** (not shown). Enter the following values**.**

- **Name** Enter an identifier for the link.
- **SIP Entity 1** Select SIP Entity defined for Session Manager during installation.
- **SIP Entity 2** Select the SIP Entity defined for the CS1000E in **Section 6.4.1.**
- **Protocol** After selecting both SIP Entities, select "**TCP**".
- **Port** Verify **Port** for both SIP entities is the default listen port.
  For the sample configuration, default listen port is "**5060**".
- **Trusted** Enter ☑
- **Notes** Enter a brief description. [Optional]

**Note**: TCP was used for the reference configuration. However, TLS would typically be used in production environments. For more information on configuring TLS, see [1] & [8].

**Step 3** - Click **Commit** to save the **Entity Link** definition.

The following screen shows the entity link defined for the SIP trunk between Session Manager and the CS1000E.




## 6.5.2 Entity Link to the Avaya Aura® SBC

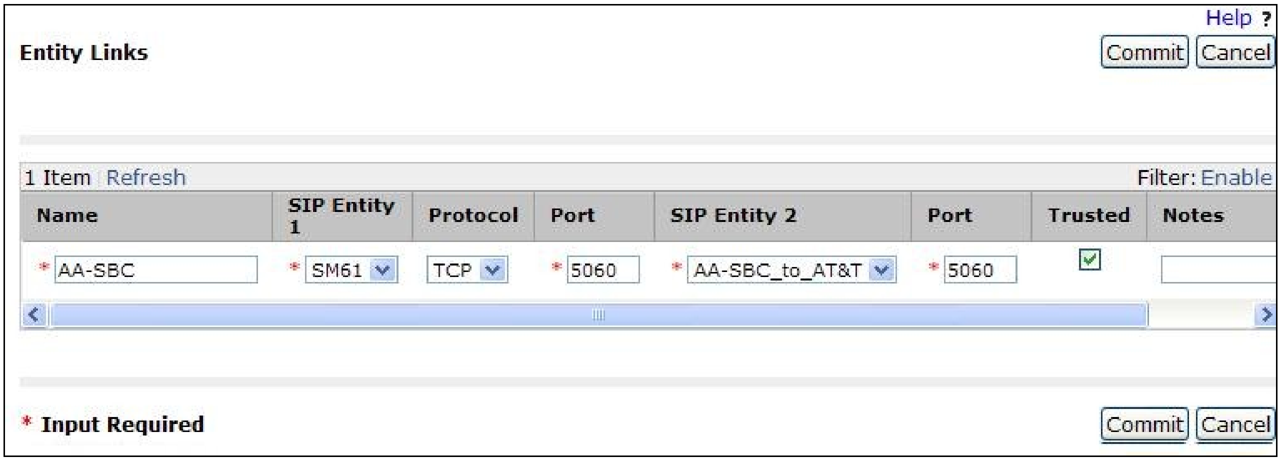
**Step 1** - Select **Entity Links** from the left navigation menu. Click **New** (not shown). Enter the following values**.**

- **Name** Enter an identifier for the link.
- **SIP Entity 1** Select SIP Entity defined for Session Manager during installation.
- **SIP Entity 2** Select the SIP Entity defined for the Avaya Aura® SBC in **Section 6.4.2**.
- **Protocol** After selecting both SIP Entities, select "**TCP**".
- **Port** Verify **Port** for both SIP entities is the default listen port. For the sample configuration, default listen port is "**5060**".
- **Trusted** Enter ☑
- **Notes** Enter a brief description. [Optional]

**Note**: TCP was used for the reference configuration. However, TLS would typically be used in production environments. For more information on TLS, see [1] & [12].

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

The following screen shows the entity link defined for the SIP trunk between Session Manager and the Avaya Aura® SBC.




## 6.6. Routing Policies

Routing policies describe the conditions under which calls will be routed by Session Manager to the CS1000E, or the Avaya Aura® SBC.

### 6.6.1 Routing Policy to the 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
- **Disabled:** Leave unchecked.
- **Notes:** Enter a brief description. [Optional]

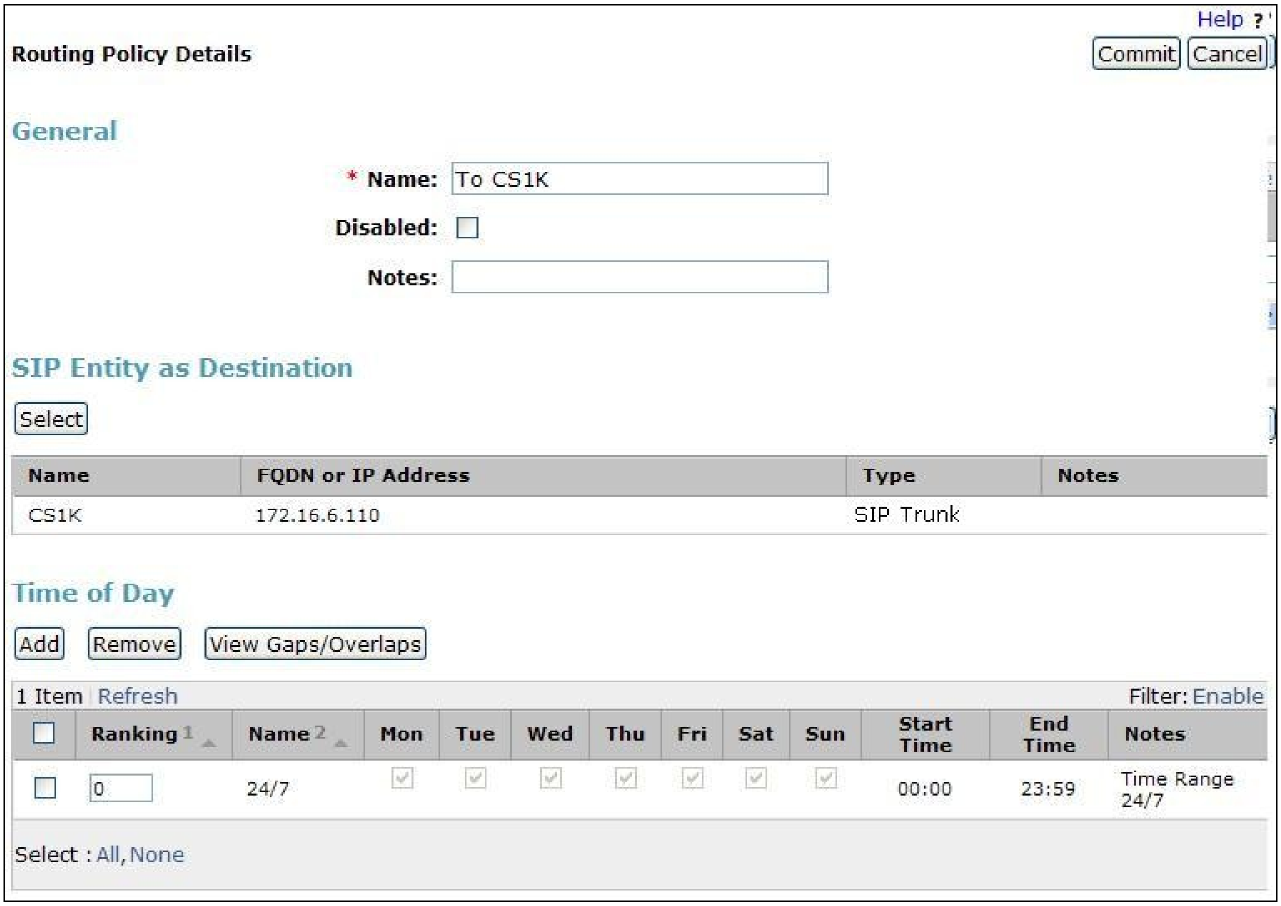
**Step 2** - In the **SIP Entity as Destination** section, click **Select.** The **SIP Entity List** page opens (not shown).

- Select the SIP Entity associated with the CS1000E (see **Section 6.4.1**) and click **Select.**
- The selected SIP Entity displays on the **Routing Policy Details** page.

**Step 3** - In the **Time of Day** section, add an appropriate time of day. 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.

The following screen shows the Routing Policy for the CS1000E.




## 6.6.2 Routing Policy to the Avaya Aura® SBC

**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
- **Disabled:** Leave unchecked.
- **Notes:** Enter a brief description. [Optional]

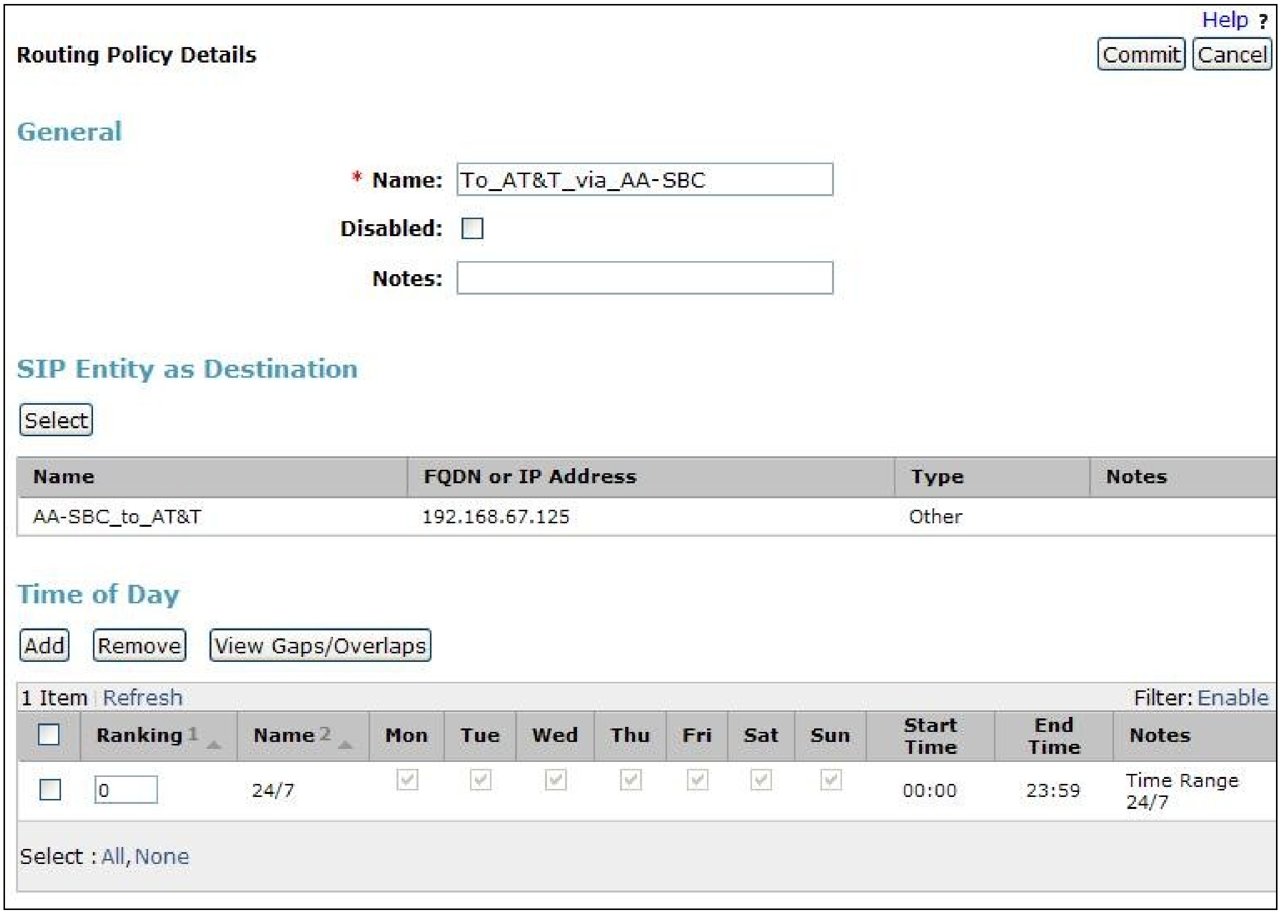
**Step 2** - In the **SIP Entity as Destination** section, click **Select.** The **SIP Entity List** page opens (not shown).

- Select the SIP Entity associated with the Avaya Aura® SBC (see **Section 6.4.2**) and click **Select.**
- The selected SIP Entity displays on the **Routing Policy Details** page.

**Step 3** - In the **Time of Day** section, add an appropriate time of day. 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.

The following screen shows the Routing Policy for the Avaya Aura® SBC.




## 6.7. Dial Patterns

Dial patterns are used to route calls to the appropriate routing policies, and ultimately to the appropriate SIP Entities. Dial patterns will be configured to route outbound calls from CS1000E users to the PSTN via the AT&T IP Flexible Reach service. Other dial patterns will be configured to route inbound calls from the AT&T IP Flexible Reach service to CS1000E users.

### 6.7.1 Inbound AT&T calls to CS1000E Users

**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.

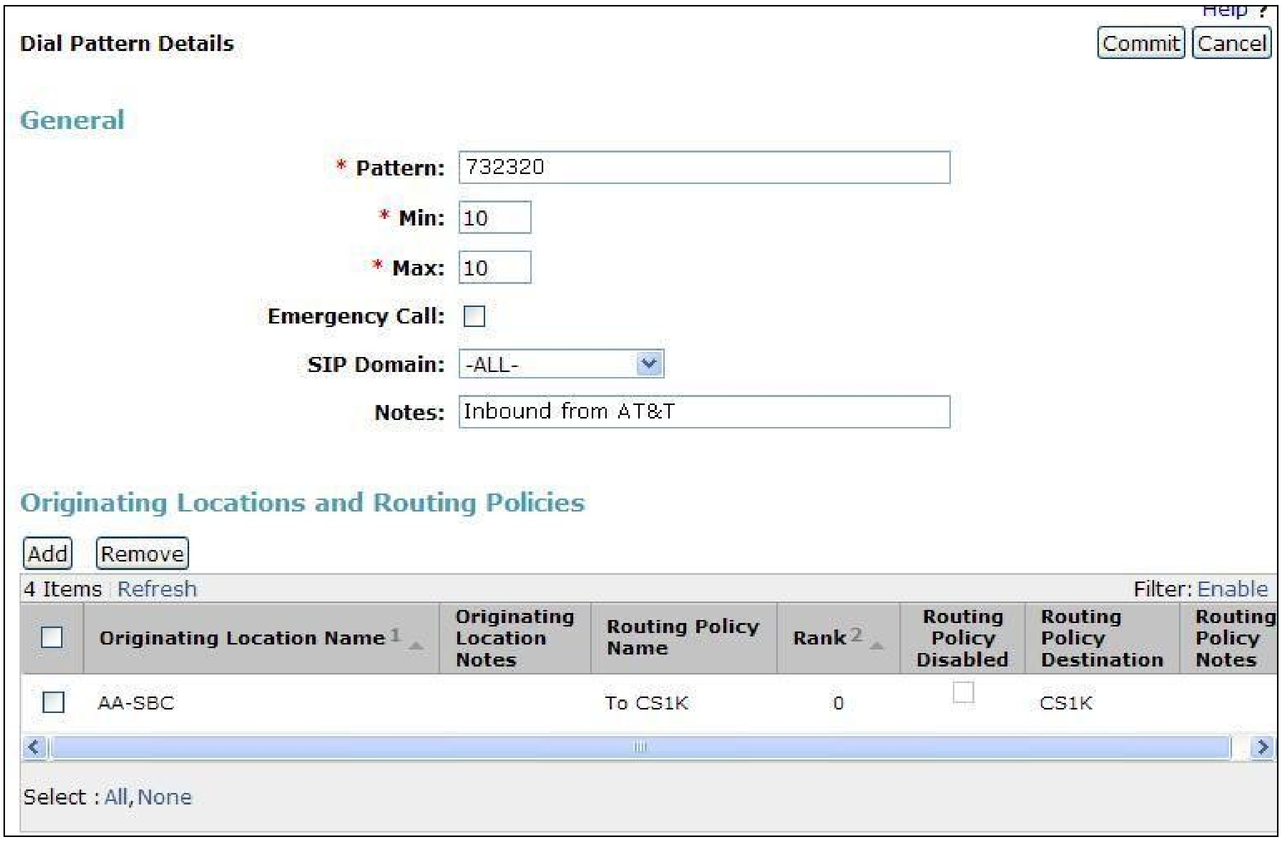
- **Pattern:** Enter dial pattern for calls to the CS1000E (e.g. 732320xxxx)
- **Min:** Enter the minimum number of digits (e.g. 10).
- **Max:** Enter the maximum number of digits (e.g. 10).
- **SIP Domain:** Select a SIP Domain from drop-down menu or select **"All"** if Session Manager should route 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 defined for the Avaya Aura® SBC in **Section 6.2.2**.
- In the **Routing Policies** table, select the Routing Policy defined for the CS1000E in **Section 6.6.1.**
- Click **Select** to save these changes and return to **Dial Pattern Details** page.

**Step 4** - Click **Commit** to save. The following screen shows an example Dial Pattern defined for the sample configuration. Repeat this procedure as needed to allow additional AT&T DID numbers to be routed to the CS1000E.




## 6.7.2 Outbound Calls to AT&T

**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.

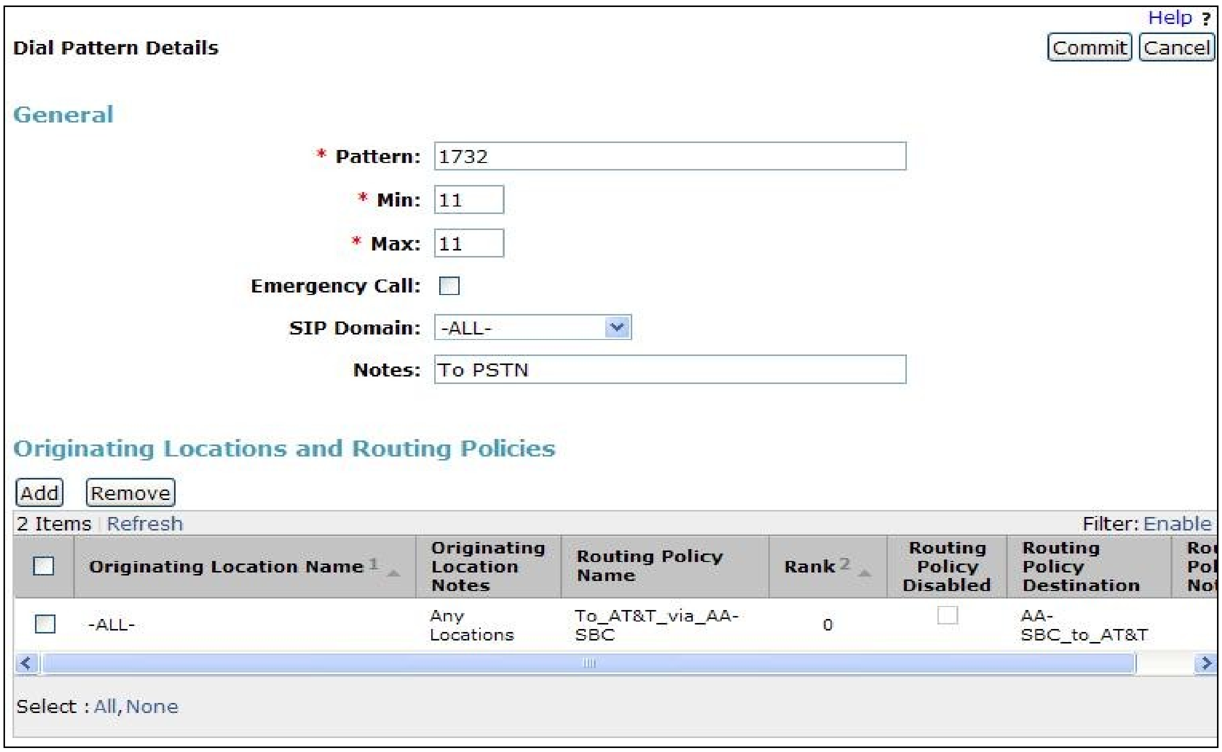
- **Pattern:** Enter dial pattern for calls destined to PSTN via the AT&T network (e.g. 1732xxxxxxx).
- **Min:** Enter the minimum number of digits (e.g. 11).
- **Max:** Enter the maximum number of digits (e.g. 11).
- **SIP Domain:** Select a SIP Domain from drop-down menu or select **"All"** if Session Manager should route outgoing 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 "**Apply the Selected Routing Policies to All Originating Locations**".  In the **Routing Policies** table, select the Routing Policy defined for the Avaya Aura® SBC in **Section 6.6.2**.
- Click **Select** to save these changes and return to **Dial Pattern Details** page.

**Step 4** - Click **Commit** to save. The following screen shows an example Dial Pattern defined for the sample configuration.  Repeat this procedure as needed to allow additional PSTN numbers to be routed to PSTN/AT&T network via the Avaya Aura® SBC.




# 7. Configure Avaya Aura® Session Border Controller (SBC)

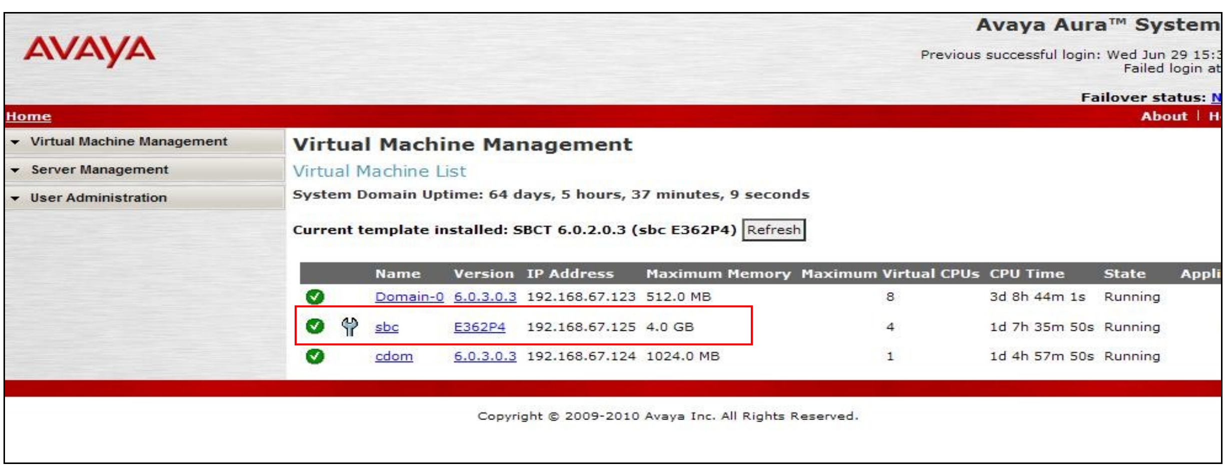
This section illustrates an example configuration of the SBC.   In the sample configuration, the SBC runs on its own S8800 Server as an application template using System Platform.  The installation of the System Platform is assumed to have been previously completed (see the Avaya Aura® SBC references in **Section 12**) for additional information on the SBC installation.

## 7.1. Logging into the Avaya Session Border Controller

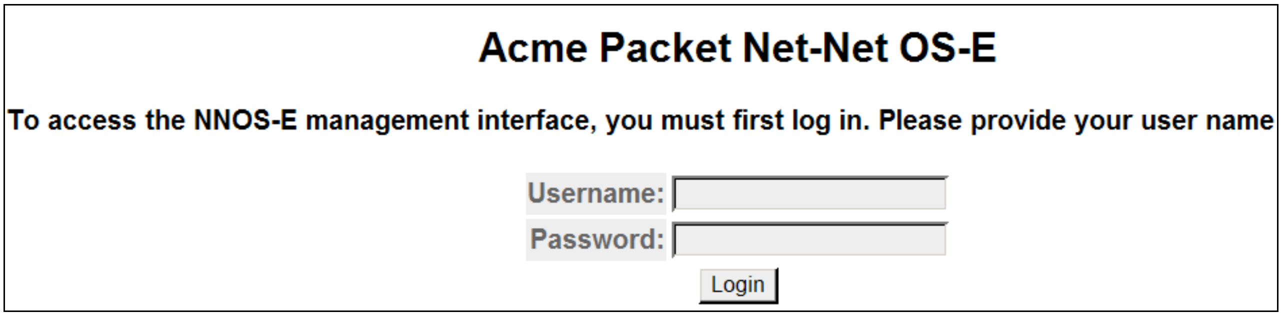
Log in to the System Platform console domain by entering https://<ip-addr>/webconsole as shown in the example screen below.  In the reference configuration, the console domain uses the IP Address 192.168.67.124.  Enter an appropriate **User Id** and press the **Continue** button.


On the subsequent screen, enter the appropriate **Password** and click the **Log On** button.

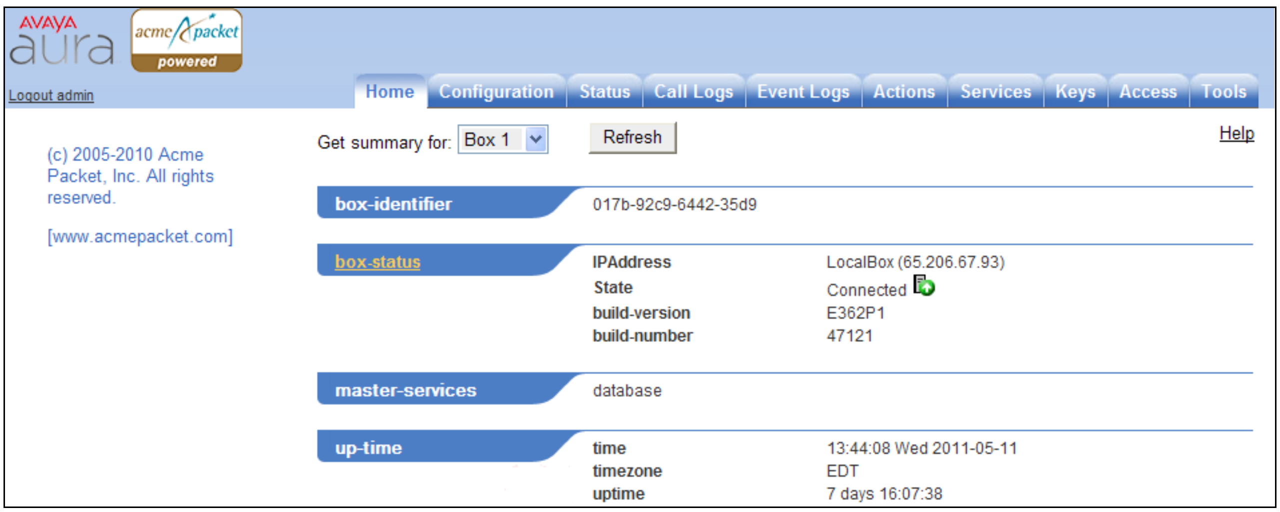

The **Virtual Machine List** will show the SBC Template installed during the SBC installation process [12] [14]. The template defines the basic SBC provisioning (IP addressing, To apply the additional reference configuration provisioning, click on the  to access the SBC GUI interface.



Enter appropriate **Username** and **Password** and click **Login**.




The following shows an abridged **Home** screen after logging in.  Note the tabs at the top.




## 7.2. Network Configuration

In the reference configuration, the Avaya S8800 Server has four physical network interfaces, labeled 1 through 4.  The port labeled “1” (virtual “eth0”) is used for the management and private (inside) network interface of the SBC (toward the customer equipment).  The port labeled “4” (virtual “eth2”) is used for the public (outside) network interface of the SBC (toward AT&T). These can be verified by checking the “interface eth0” and interface eth2” settings (see **Section 7.2.1**).

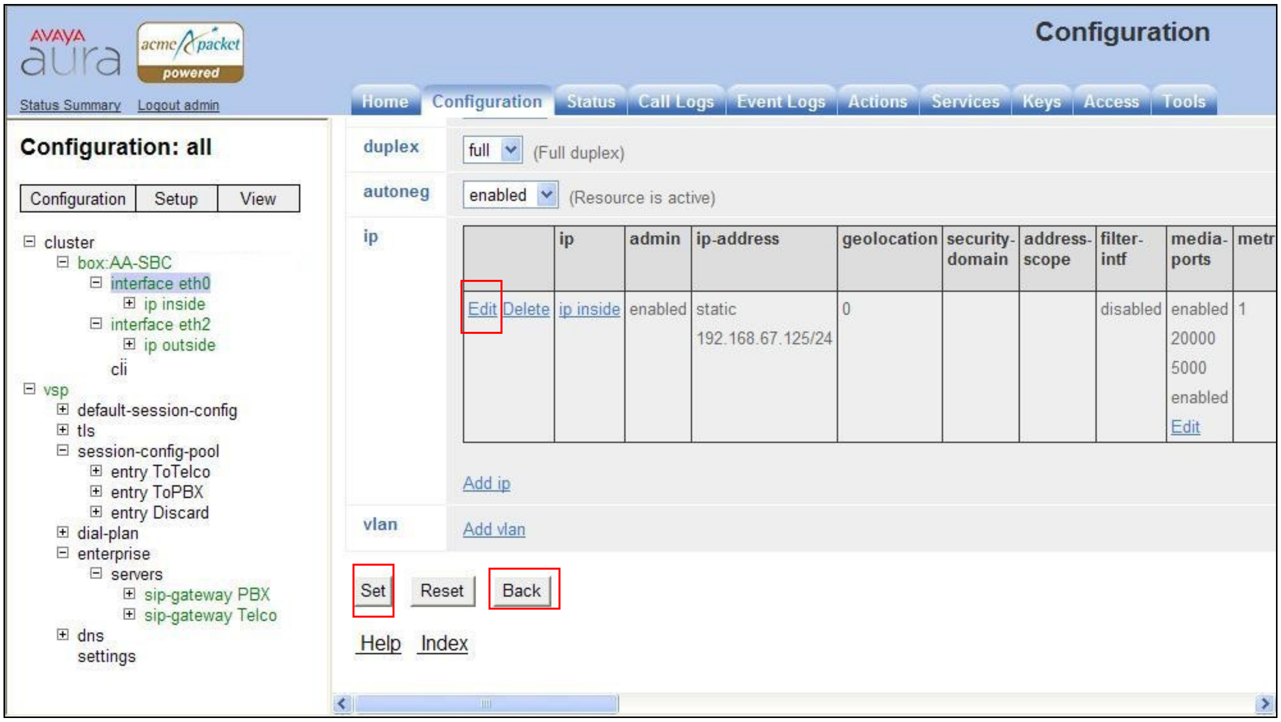
The AT&T AVPN transport service requires that RTP media traffic use UDP port range 16384-32767. This range is defined as part of “interface eth2” (see **Section 7.2.3**).

SIP-Gateways are defined for corresponding to the private and public interfaces. In the reference configuration the private interface is defined as “PBX” and the public interface is defined as “Telco1” (see **Section 7.2.4**).

### 7.2.1 Verify IP Addressing

**Step 1** - From the **Configuration** tab, select **cluster → box <name defined during install>** (e.g. **AA-SBC**). The **interface eth0** and **interface eth2** will be displayed. Click on **ip inside** (eth0) or **ip outside** (eth2) to display the interface configuration. Note that AT&T may require the eth2 IP address as part of the IP Flexible Reach service provisioning.

**Step 2** - The configuration may be modified by clicking the **Edit** button. If changes are made, click on the **Set** button. To cancel changes or to go to a previous screen, click on **Back**.



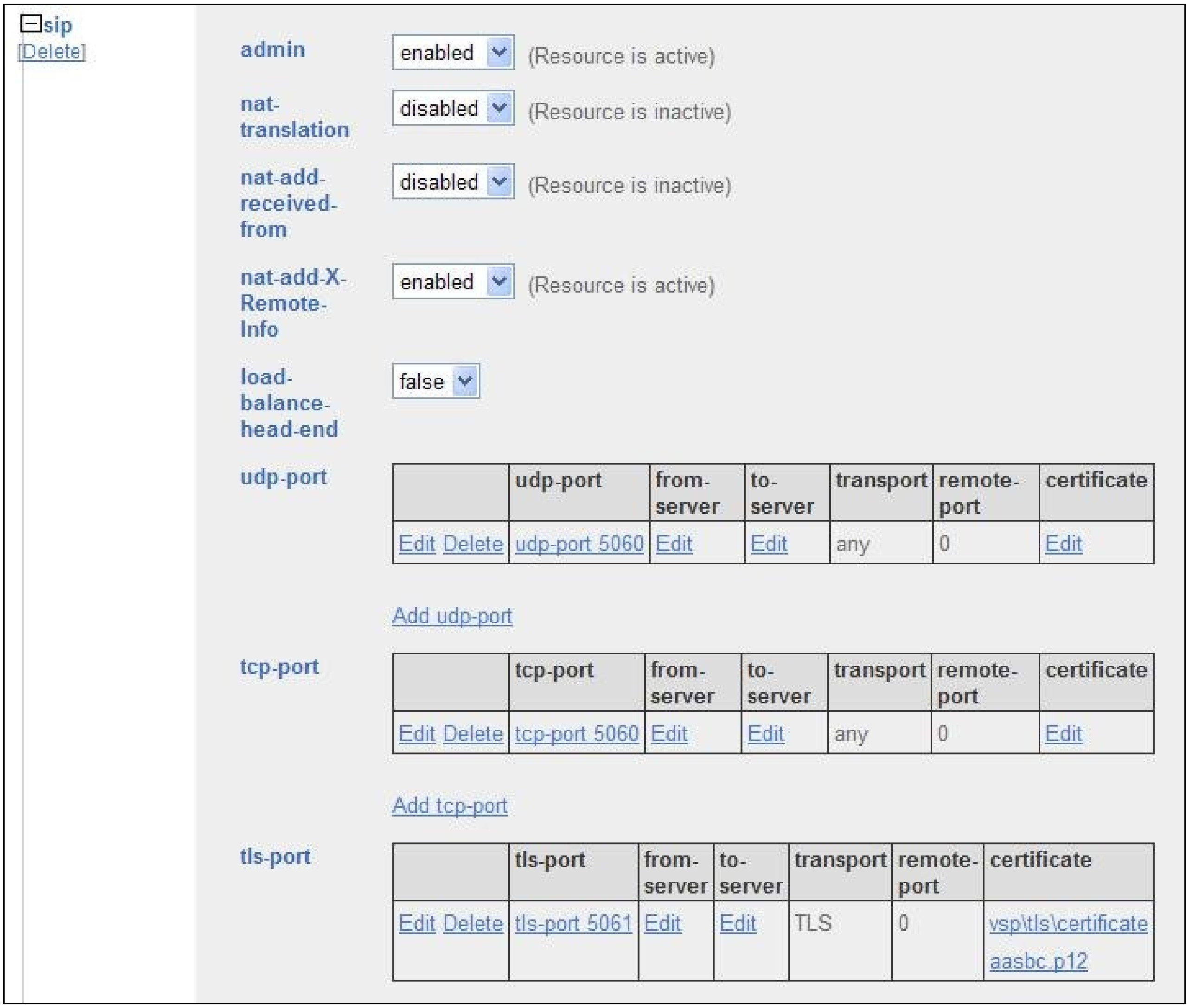

### 7.2.2 Transport Protocols

#### 7.2.2.1 Private Interface – Eth0

The private interface, eth0, was provisioned to support UDP, TCP, and TLS transport protocols. However, TCP (port 5060) was used in the reference configuration for the connection to Session Manager (see **Section 6.5.2**). This can be displayed by the following:

**Step 1** – Navigate to **cluster → box <name defined during install> → interface eth0 → ip inside.**

**Step 2 –** Scroll down to, and click on the **SIP** heading. The UDP, TCP, and TLS supported protocols are displayed.
**Note**: TCP was used for the reference configuration. However, TLS would typically be used in production environments. For more information on configuring TLS, see [12].



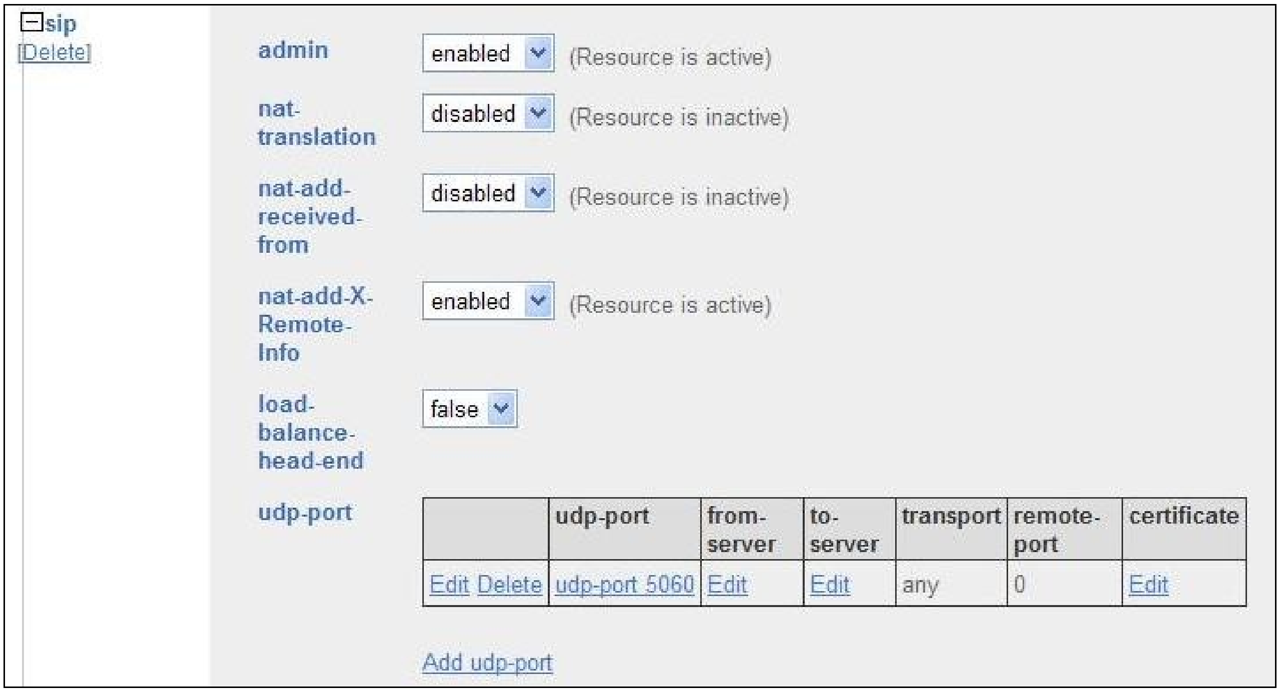

**Step 3** - The configuration may be modified by clicking the **Edit** buttons. If changes are made, click on the **Set** button (not shown). To cancel changes or to go to a previous screen, click on **Back** (not shown).

#### 7.2.2.2 Public Interface – Eth2

The AT&T IP Flexible Reach service requires UDP transport protocol between the Avaya Aura® SBC and the AT&T IP Flexible Reach service border element. Therefore, the public interface, eth2, was provisioned to support UDP transport protocol only. This can be displayed by the following:

**Step 1** – Navigate to **cluster → box <name defined during install> → interface eth2 → ip outside.**

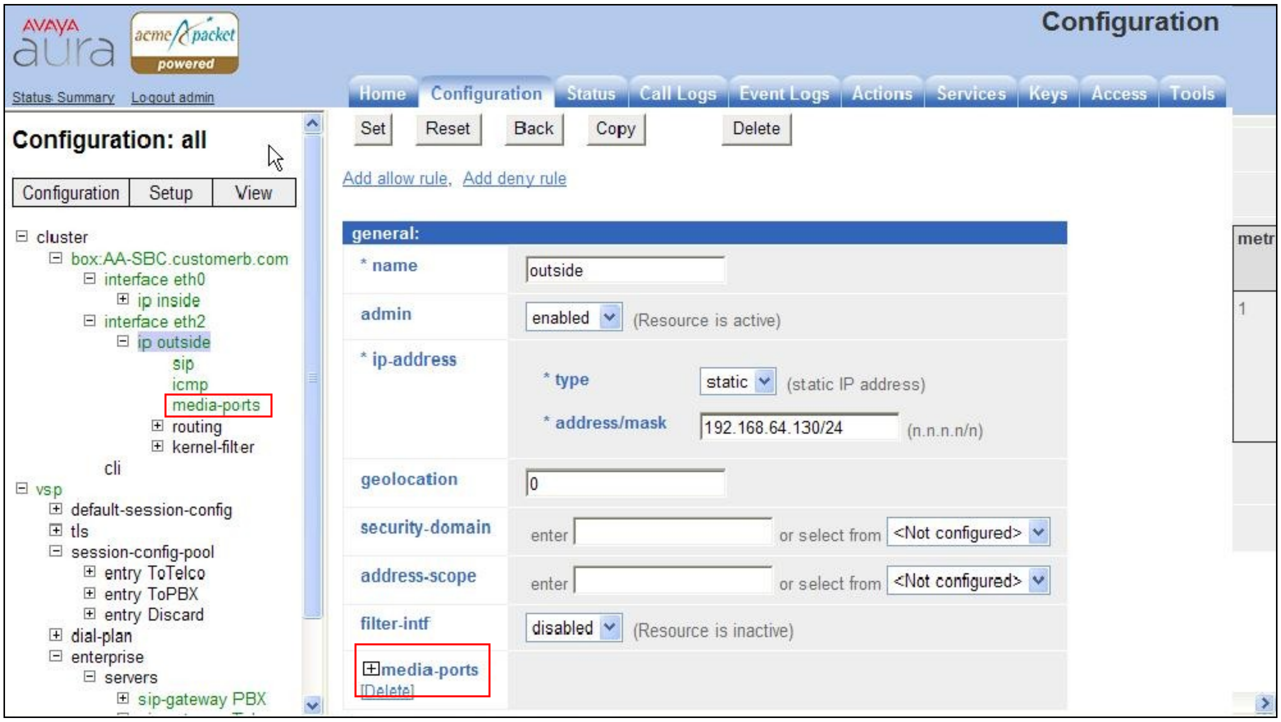
**Step 2 –** Scroll down to, and click on the **SIP** heading. The UDP (port 5060) transport protocol is displayed.



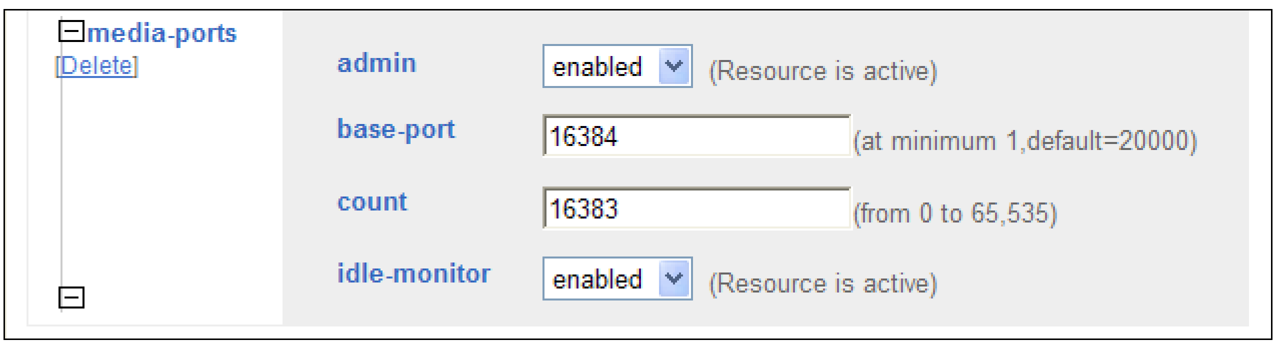

**Step 3** - The configuration may be modified by clicking the **Edit** buttons. If changes are made, click on the **Set** button (not shown). To cancel changes or to go to a previous screen, click on **Back** (not shown).

### 7.2.3 Setting the RTP Port Range on Eth2

**Step 1** - Go to **cluster → box <name defined during install> → interface eth2 → ip outside** to display the eth2 configuration toward AT&T. Select Media Ports from either the menu or from the display.



**Step 2** - The media port section will be displayed. Enter **16384** in the **base-port** field and **16383** in the **count** field.



**Step 3** - Click on the **Set** button to save.

**Step 4** - Proceed to save and activate the configuration as described in **Section 7.3**.

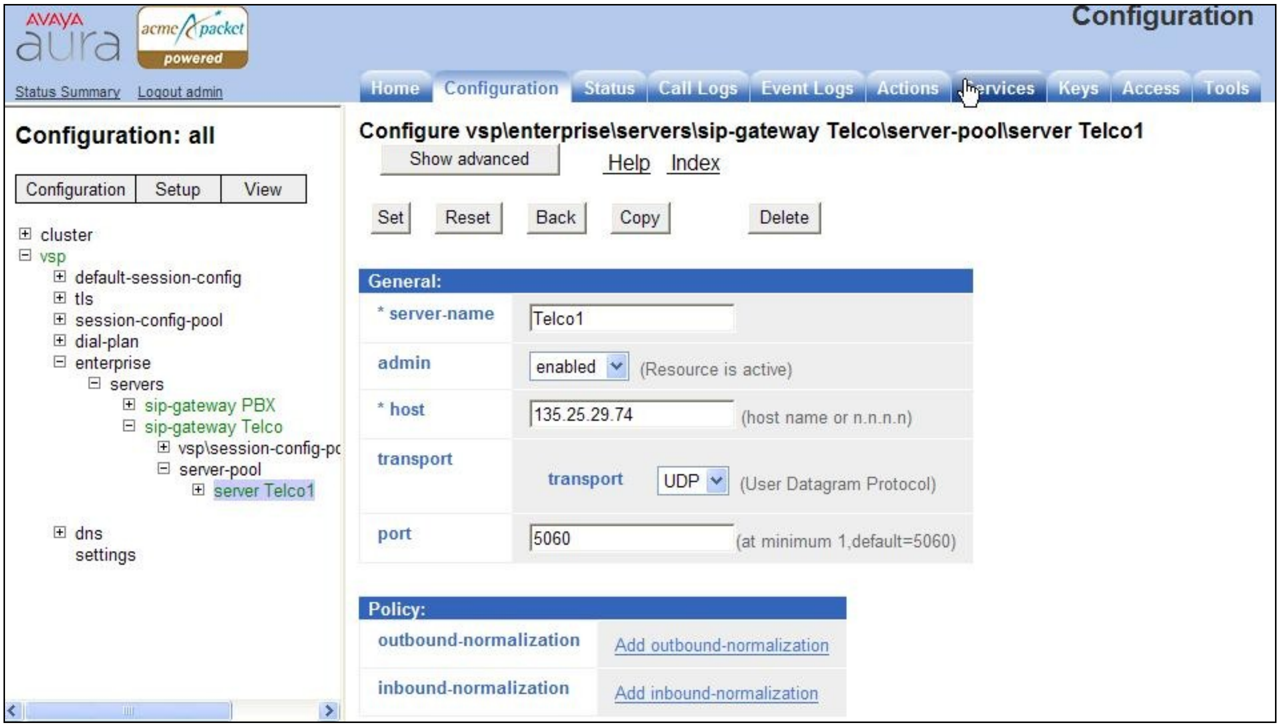
### 7.2.4 Configuring the SIP-Gateways

In the reference configuration, a sip-gateway was defined to AT&T (the IP Flexible Reach border element) and to the customer site (Session Manager). The AT&T gateway was defined as "Telco1" and customer gateway was defined as "PBX".

#### 7.2.4.1 Telco1

**Step 1** - Go to **vsp → enterprise → servers** and any previously defined sip-gateways will be displayed. In the reference configuration sip-gateways **PBX** and **Telco1** were defined.

**Step 2** - Click on **sip-gateway Telco → servers → server-pool → server Telco1** and the Telco1 sip-gateway configuration will be displayed.




**Step 3** - Verify the following:

- admin state is **enabled**.
- host address is the IP address of the AT&T IP Flexible Reach border element (e.g. **135.25.29.74**).
- transport protocol is **UDP**.
- port is **5060**.

**Step 4** - Click on the **Set** button to save any changes or **Back** if no changes are required.

**Step 5** - Proceed to save and activate the configuration as described in **Section 7.3**.

#### 7.2.4.2 PBX

Repeat the steps in **Section 7.2.4.1** and verify the following:

- admin state is **enabled**.
- host address is the IP address of Session Manager (e.g. **192.168.67.210**).
- transport protocol is **TCP**. Note that TCP was used in the reference configuration to facilitate protocol trace verification and troubleshooting. TLS may be used as well (see [1] & [12].
- port is **5060**.

### 7.2.5 Stripping SIP Headers

The Avaya Aura® SBC can be used to strip SIP headers that are not required or supported by AT&T.  For headers that have relevance only within the enterprise, it may be desirable to prevent the header from being sent to the public SIP Service Provider. For example, Session Manager Release 6.1 may insert the "P-Location" and "Remote-Party-ID" headers. The CS1000E may send the "x-nt-e164-clid", "x-nt-corr-id", "Alert-Info", and "History-Info" headers. The following procedures may be used to strip such headers that AT&T does not process.

#### 7.2.5.1 Specific Session-config-pool Method

Undesired headers may be removed via the session-config-pool. For example, during installation, two session-config-pools were created, "To-Telco" and "To-PBX". First the headers are removed session-config-pool "**To-Telco**". This will remove the specified headers for calls sent by the customer location to AT&T.

**Step 1** - Navigate to **vsp → session-config-pool → entry ToTelco → header-settings**. In the resultant screen, click **Edit blocked-header** and proceed to add the P-Location and other blocked headers as described in the previous section.


**Step 2 –** Repeat the procedure in **Step 1** for **"entry ToPBX".** This will remove the specified headers for calls from AT&T and answered by the customer location.

**Step 3** - Proceed to save and activate the configuration as described in **Section 7.3**.

### 7.2.6 Stripping Unnecessary SIP Message Body Information

As described in **Section 6.3.2**, the message body of an INVITE message sent from the CS1000E will contain a MIME Multipart message body headers not required by AT&T. These headers were removed by Session Manager in the reference configuration. Alternatively the Avaya Aura® SBC may be used to remove these headers. Two alternative approaches were tested successfully and are described below as reference. In one approach, the SBC is used to specifically block the "x-nt-mcdn-frag-hex" and "x-nt-epid-frag-hex" parts. In another approach, the SBC is used to block any body part that is not SDP.

#### 7.2.6.1 Block Any body part but SDP Approach

**Step 1** - To block any body part but SDP, navigate to **vsp → default-session-config → bodypart-type**. Click **Add allowed-body-part**.

**Step 2** - In the **bodypart-type** drop-down menu, select "application". In the application-sub-type menu, select "sdp" as shown in the screen below. Click **Create**.

**Step 3** - Navigate to **vsp → default-session-config → bodypart-type**. Click **Add blocked-body-part**.

**Step 4** - In the **bodypart-type** drop-down menu, select "**application**". In the application-sub-type menu, select "**any**" as shown in the screen below. Click **Create**.

**Step 5** - Proceed to save and activate the configuration as described in **Section 7.3**.

#### 7.2.6.2 Block Specific Body Part Approach

This is an alternative to the approach documented in the previous sub-section. In this section, the specific body parts that the CS1000E inserts in the message body are blocked rather than blocking anything but SDP.

**Step 1** - Navigate to **vsp → default-session-config → bodypart-type**. Click **Add blocked-body-part**.

**Step 2** - In the **bodypart-type** drop-down menu, select "**application**". In the **application-sub-type** menu, type in or select "**x-nt-mcdn-frag-hex**". Click **Create**.

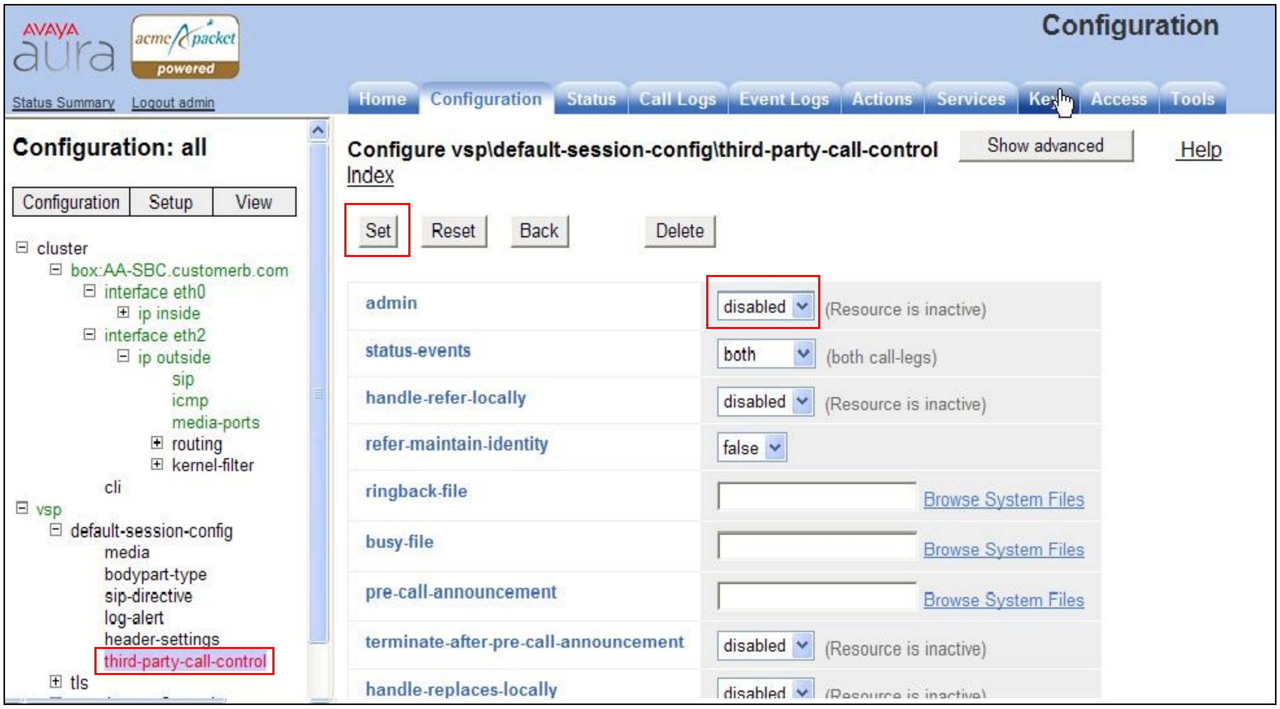
**Step 3** - Click **Add blocked-body-part.** In the **bodypart-type** drop-down menu, select "**application**". In the **application-sub-type** menu, type in or select "**x-nt-epid-frag-hex**". Click **Create**.

**Step 4** - Proceed to save and activate the configuration as described in **Section 7.3**.

### 7.2.7 Disable Third Party Call Control

**Step 1** - Navigate to **vsp → default-session-config → third-party-call-control**. To disable third-party-call-control, select **disabled** from the **admin** drop-down. Note - After disabling, the third-party-call-control link becomes red as shown below.

**Step 2** - click **Set** as shown below.



**Step 3** - Proceed to save and activate the configuration as described in **Section 7.3**.

### 7.2.8 SIP OPTIONS Messages for AT&T Network Status

In the reference configuration, the Avaya Aura® SBC sent SIP OPTIONS messages to the AT&T IP Flexible Reach border element to verify the state of the network connection. The AT&T response to the OPTIONS is "405 Method Not Allowed". Although this appears to be an error, in fact the arrival of the message assures the Avaya Aura® SBC that the network connection is up.

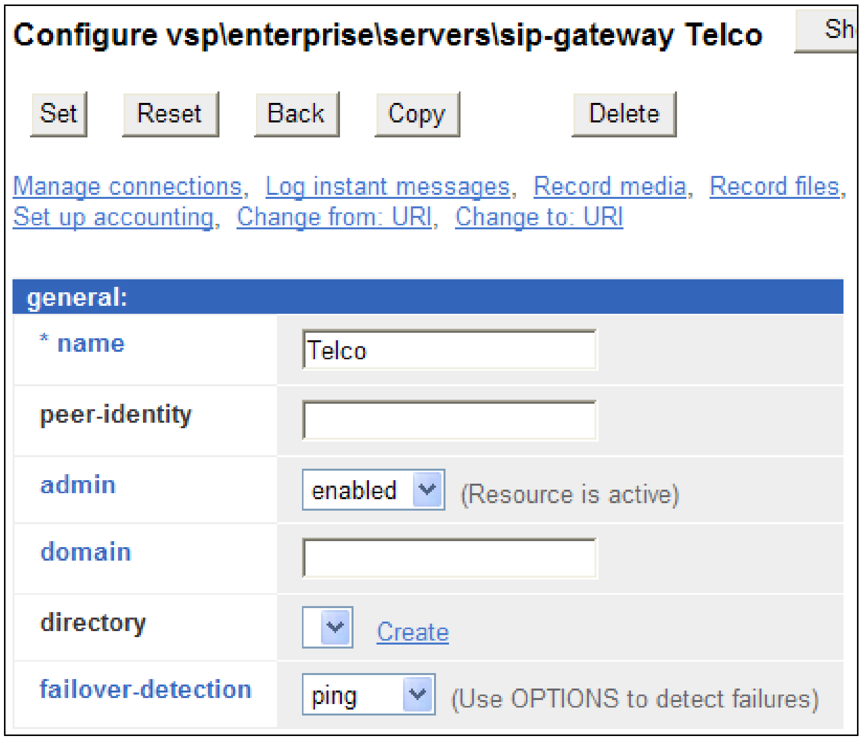
**Step 1** - Navigate to **cluster → box:AvayaSBC → interface eth2 → ip outside**. Scroll down to, and click on, the **icmp** option.

**Step 2** - Set the **admin** option to **enabled**.

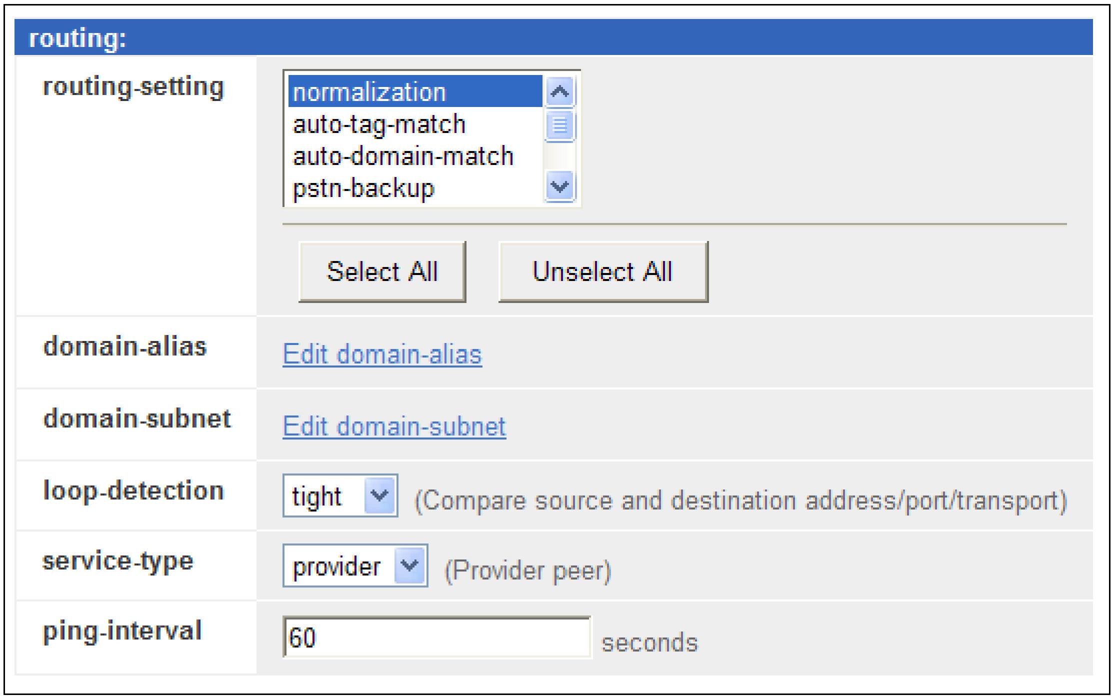
**Step 3** - Scroll to the bottom of the screen and click **Set**.

**Step 4** - Navigate to **vsp → enterprise → servers → sip-gateway Telco**. Click on the **Show Advanced** button at the top of the page (not shown).

**Step 5** – In the **general:** section set **failover-detection** and select **ping** from the menu.




**Step 6** – Scroll down to the **routing**: section and set the **ping-interval** as desired (e.g. **60**).




**Step 7** - Scroll to the bottom of the screen and click **Set**.

**Step 8** - Proceed to save and activate the configuration as described in **Section 7.3**.

### 7.2.9 Altering the To Header for PSTN Calls to CS1000E

The AT&T IP Flexible Reach service may specify different DID numbers in the Request URI and To headers of inbound Invite messages. The Avaya Aura® SBC is configured to modify the To header to match the number (user field) specified in the Request URI.

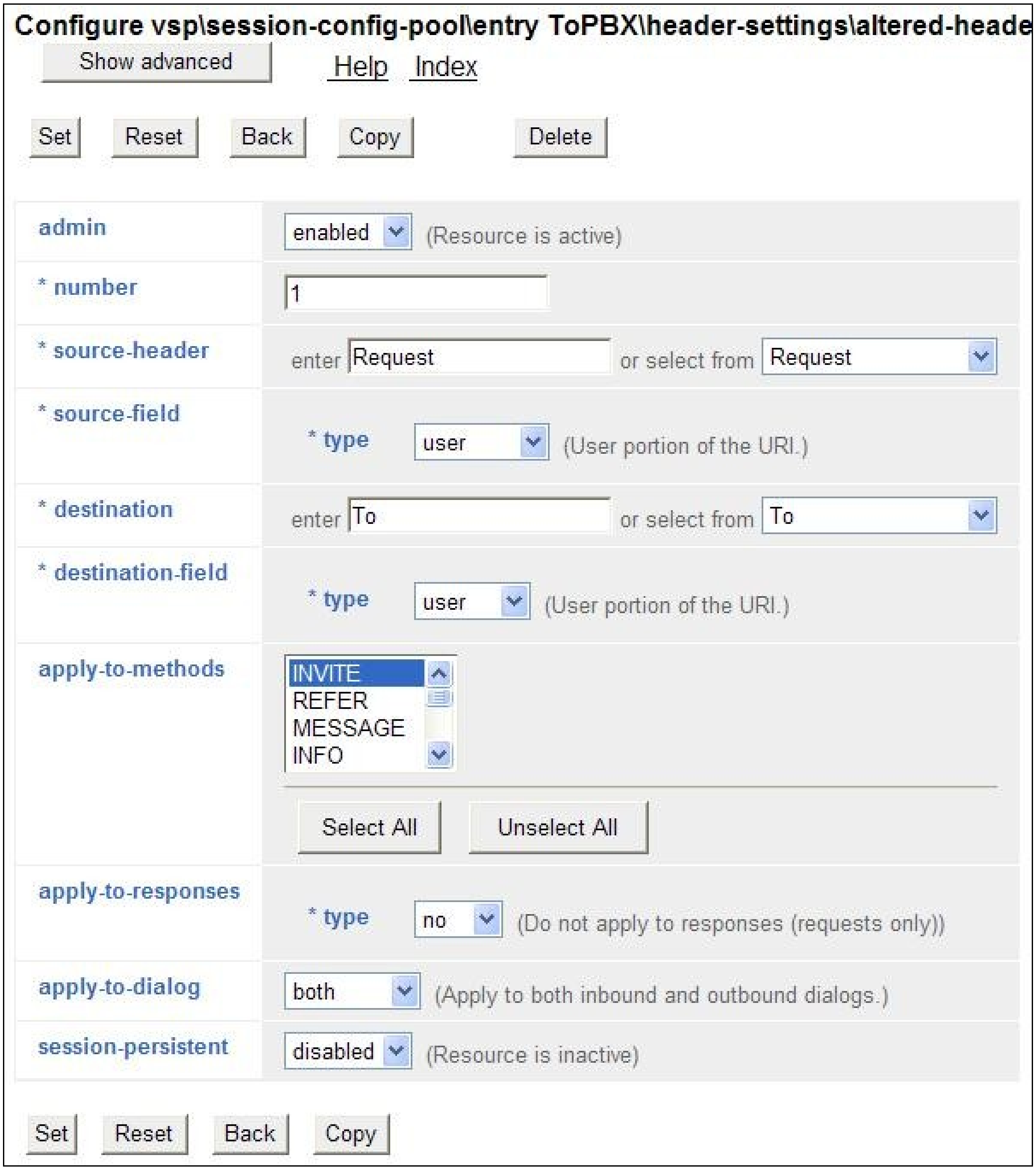
| |
|---|
| **Note** - While not required for typical inbound calls, this function is required for correct handling of inbound calls to Call Pilot for message retrieval. |

**Step 1** - Navigate to **vsp → session-config-pool → entry ToPBX → header-settings → altered-header**, and click on **Add altered –header** (not shown). In the resultant screen, enter the follow:

- **admin** – **enabled**
- **number** – select an identifying number (e.g. **1**).
- **source-header** – Select **Request** from the drop down menu.
- **source-field** – Select **User** from the drop down menu.
- **destination** – Select **To** from the drop down menu.
- **destination-field** - Select **User** form the drop down menu.
- **apply-to-methods** – Select **INVITE** from the drop down menu.
- Let the other values default

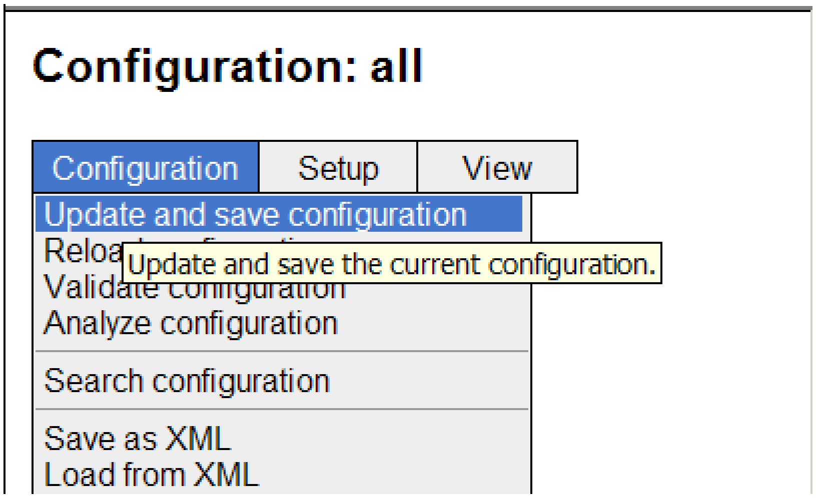
**Step 2** - Scroll to the bottom of the screen and click **Set**.

**Step 3** - Proceed to save and activate the configuration as described in **Section 7.3**.

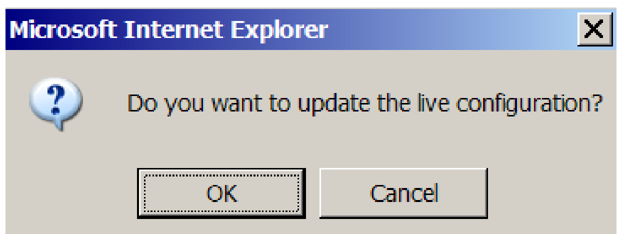



## 7.3. Saving and Activating Configuration Changes

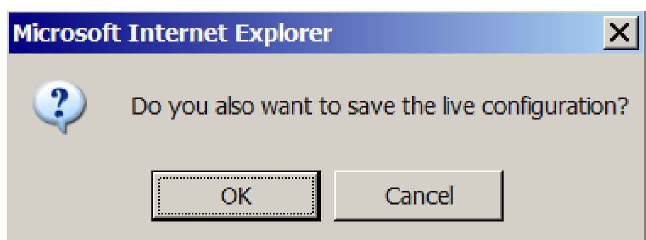
**Step 1** - To save and activate configuration changes, select **Configuration → Update and save configuration** from the upper left hand side of the user interface, as shown below.



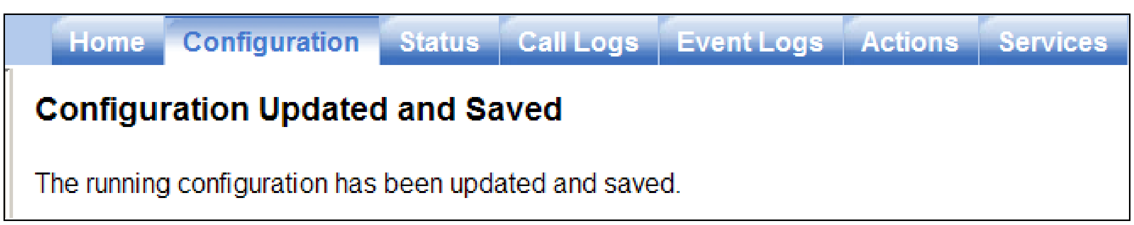

**Step 2** - Click **OK** to update the live configuration.




**Step 3** - Click **OK** to save the live configuration.




A screen that includes the following should appear.




## 7.4. Avaya Aura® SBC Configuration File

The Avaya Aura® SBC configuration is saved to a text configuration file (**cxc.cfg** file). A copy of the configuration file can be retrieved from the SBC by selecting the **Tools** tab and selecting

**Download saved configuration file** from the left-side menu. An example configuration file resulting from the configuration in **Section 7** is attached.

```
#
#  Copyright (c) 2004-2011  Acme Packet Inc.
#  All Rights Reserved.
#
#  File: /cxc/cxc.cfg
#  Date: 13:53:59 Wed 2011-08-03
#
config cluster
 config box 1
  set hostname AA-SBC.customerb.com
  set timezone America/New_York
  set name AA-SBC.customerb.com
  set identifier 00:ca:fe:45:93:63
  config interface eth0
   config ip inside
    set ip-address static 192.168.67.125/24
    config ssh
    return
    config snmp
     set trap-target 192.168.67.124 162
     set trap-filter generic
     set trap-filter dos
     set trap-filter sip
     set trap-filter system
    return
    config web
    return
    config web-service
     set protocol https 8443
     set authentication certificate "vsp\tls\certificate ws-cert"
    return
    config sip
     set udp-port 5060 "" "" any 0
     set tcp-port 5060 "" "" any 0
     set tls-port 5061 "" "" TLS 0 "vsp\tls\certificate aasbc.p12"
    return
    config icmp
    return
    config media-ports
    return
    config routing
     config route Default
      set gateway 192.168.67.1
     return
     config route Static0
      set destination network 192.11.13.4/30
      set gateway 192.168.67.123
     return
     config route Static1
      set admin disabled
     return
     config route Static2
      set admin disabled
```

```
    return
    config route Static3
     set admin disabled
    return
    config route Static4
     set admin disabled
    return
    config route Static5
     set admin disabled
    return
    config route Static6
     set admin disabled
    return
    config route Static7
     set admin disabled
    return
   return
  return
 return
 config interface eth2
  config ip outside
   set ip-address static 192.168.64.130/24
   config sip
    set udp-port 5060 "" "" any 0
   return
   config icmp
   return
   config media-ports
    set base-port 16384
    set count 16383
   return
   config routing
    config route Default
     set admin disabled
    return
    config route external-sip-media-1
     set destination network 135.25.29.0/24
     set gateway 192.168.64.254
    return
    config route ORT
     set destination network 12.40.234.0/24
     set gateway 192.168.64.254
    return
   return
   config kernel-filter
    config allow-rule allow-sip-udp-from-peer-1
     set destination-port 5060
     set source-address/mask 135.25.29.0/24
     set protocol udp
    return
    config deny-rule deny-all-sip
     set destination-port 5060
    return
   return
  return
 return
 config cli
```

```
   set prompt AA-SBC.customerb.com
  return
 return
return

config services
 config event-log
  config file access
   set filter access info
   set count 3
  return
  config file system
   set filter system info
   set count 3
  return
  config file errorlog
   set filter all error
   set count 3
  return
  config file db
   set filter db debug
   set filter dosDatabase info
   set count 3
  return
  config file management
   set filter management info
   set count 3
  return
  config file peer
   set filter sipSvr info
   set count 3
  return
  config file dos
   set filter dos alert
   set filter dosSip alert
   set filter dosTransport alert
   set filter dosUrl alert
   set count 3
  return
  config file krnlsys
   set filter krnlsys debug
   set count 3
  return
 return
return

config master-services
 config database
  set media enabled
 return
return

config vsp
 set admin enabled
 config default-session-config
  config media
   set anchor enabled
```

```
  set rtp-stats enabled
 return
 config bodypart-type
 return
 config sip-directive
  set directive allow
 return
 config log-alert
 return
 config header-settings
 return
 config third-party-call-control
  set handle-refer-locally disabled
 return
return
config tls
 config default-ca
  set ca-file /cxc/certs/sipca.pem
 return
 config certificate ws-cert
  set certificate-file /cxc/certs/ws.cert
 return
 config certificate aasbc.p12
  set certificate-file /cxc/certs/aasbc.p12
  set passphrase-tag aasbc-cert-tag
 return
return
config session-config-pool
 config entry ToTelco
  config to-uri-specification
   set host next-hop
  return
  config from-uri-specification
   set host local-ip
  return
  config request-uri-specification
   set host next-hop
  return
  config p-asserted-identity-uri-specification
   set host local-ip
  return
  config contact-uri-settings-in-leg
  return
  config contact-uri-settings-out-leg
  return
  config bodypart-type
   set blocked-body-part custom-mime-type x-nt-inforeq any
  return
  config in-codec-preferences
   set preference audio "Media Format: DynamicRTP-Type-111" 0
  return
  config out-codec-preferences
   set preference audio "Media Format: DynamicRTP-Type-111" 0
  return
  config out-media-normalization
  return
  config header-settings
```

```
    set blocked-header P-Location
    set blocked-header x-nt-e164-clid
    set blocked-header x-nt-corr-id
    set blocked-header Alert-Info
    set blocked-header History-info
    set blocked-header Remote-Party-ID
    config reg-ex-header 1
     set admin disabled
     set destination From
     set create From "<sip:\+(.*)@(.*)" "<sip:\1@\2"
    return
    config reg-ex-header 2
     set admin disabled
     set destination P-Asserted-Identity
     set create P-Asserted-Identity "<sip:\+(.*)@(.*)" "<sip:\1@\2"
    return
    config reg-ex-header 3
     set admin disabled
     set destination Contact
     set create Contact "<sip:\+(.*)@(.*)" "<sip:\1@\2"
    return
   return
  return
  config entry ToPBX
   config to-uri-specification
    set host next-hop-domain
   return
   config request-uri-specification
    set host next-hop-domain
   return
   config contact-uri-settings-in-leg
   return
   config contact-uri-settings-out-leg
   return
   config in-codec-preferences
    set preference audio "Media Format: DynamicRTP-Type-111" 0
   return
   config out-codec-preferences
    set preference audio "Media Format: DynamicRTP-Type-111" 0
   return
   config header-settings
    set blocked-header P-Location
    set blocked-header x-nt-e164-clid
    set blocked-header x-nt-corr-id
    set blocked-header Alert-Info
    set blocked-header History-Info
    set blocked-header Remote-Party-ID
    config altered-header 1
     set source-header Request
     set source-field user
     set destination To
     set destination-field user
    return
   return
  return
  config entry Discard
   config sip-directive
```

```
    return
   return
  return
  config dial-plan
   config route Default
    set priority 500
    set location-match-preferred exclusive
    set session-config vsp\session-config-pool\entry Discard
   return
   config source-route FromTelco
    set peer server "vsp\enterprise\servers\sip-gateway PBX"
    set source-match server "vsp\enterprise\servers\sip-gateway Telco"
   return
   config source-route FromPBX
    set peer server "vsp\enterprise\servers\sip-gateway Telco"
    set source-match server "vsp\enterprise\servers\sip-gateway PBX"
   return
  return
  config enterprise
   config servers
    config sip-gateway PBX
     set domain cots1.ntlab.com
     set failover-detection ping
     set outbound-session-config-pool-entry vsp\session-config-pool\entry ToPBX
     config server-pool
      config server PBX1
       set host 192.168.67.210
       set transport TCP
      return
     return
    return
    config sip-gateway Telco
     set failover-detection ping
     set ping-interval 60
     set outbound-session-config-pool-entry vsp\session-config-pool\entry
ToTelco
     config server-pool
      config server Telco1
       set host 135.25.29.74
       set connection-retry-interval 60
       config network
       return
       config error-response-codes
       return
      return
      config server Telco2
       set admin disabled
       set host 135.25.29.75
       config network
       return
       config error-response-codes
       return
      return
      config server ORT
       set admin disabled
       set host 12.40.234.99
      return
```

```
    return
   return
  return
 return
 config dns
  config resolver
   config server 192.168.67.5
   return
  return
 return
 config settings
  set read-header-max 8191
 return
return

config external-services
return

config preferences
 config gui-preferences
  set enum-strings SIPSourceHeader PAI
  set enum-strings SIPSourceHeader P-Asserted-Identity
  set enum-strings SIPSourceHeader Contact
  set enum-strings MediaAudioSubType "Media Format: DynamicRTP-Type-111"
 return
return

config access
 config permissions superuser
  set cli advanced
 return
 config permissions read-only
  set config view
  set actions disabled
 return
 config users
  config user admin
   set password 0x00b930c8c97cc6705c312dd835419ecc3559106a7b9f91774cb86e85ec
   set permissions access\permissions superuser
  return
  config user cust
   set password 0x0033a56c33e6e62e159bb5bd94be32dc30e408d441627c93d9d740483c
   set permissions access\permissions read-only
  return
  config user init
   set password 0x002e3afdb5919e72cbd542345a7a918f9cad4ce7c917bcff336fc9901b
   set permissions access\permissions superuser
  return
  config user craft
   set password 0x00fb8b12eba46bc122cf5642e1e076477a510ffa51da44498020fcbc12
   set permissions access\permissions superuser
  return
  config user dadmin
   set password 0x0005b4d2ba1868181287ff79c199ab43f8575ca330d8d88599dde14804
   set permissions access\permissions read-only
  return
 return
```

```
return

config features
return
```

# 8. AT&T IP Flexible Reach Service

Information regarding AT&T IP Flexible Reach Service may be found at http://www.business.att.com/enterprise/Service/voice-services/voip/sip-trunking/ or by contacting AT&T at **800-248-3632**.

## 8.1. AT&T Provisioning

The AT&T IP Flexible Reach service provided DID numbers for the reference configuration that could be called from the PSTN. These DID numbers terminated to the Avaya CS1000E location via the AT&T IP Flexible Reach service. DID numbers shown in these application notes are examples. Customers will be assigned DIDs by AT&T.

The AT&T IP Flexible Reach service also provided a network border element IP address for the reference configuration. Customers will be assigned a border element IP address(es) by AT&T.

# 9. Verification Steps

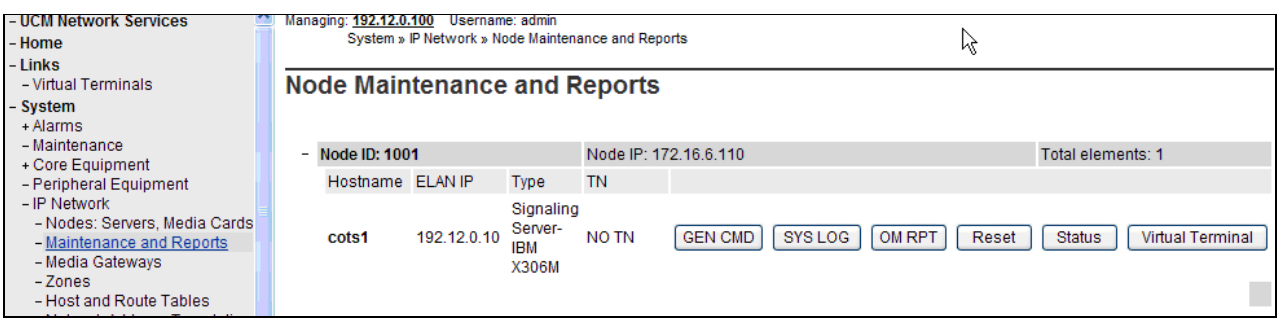
This section provides example verifications of the Avaya configuration with AT&T IP Flexible Reach service.

## 9.1. Avaya CS1000E Verifications

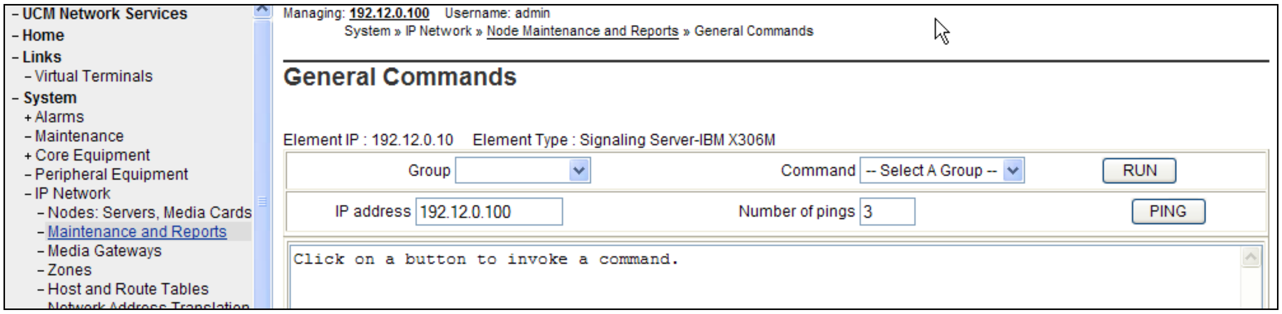
This section illustrates sample verifications that may be performed using the Avaya CS1000E Element Manager GUI.

### 9.1.1 IP Network Maintenance and Reports Commands

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

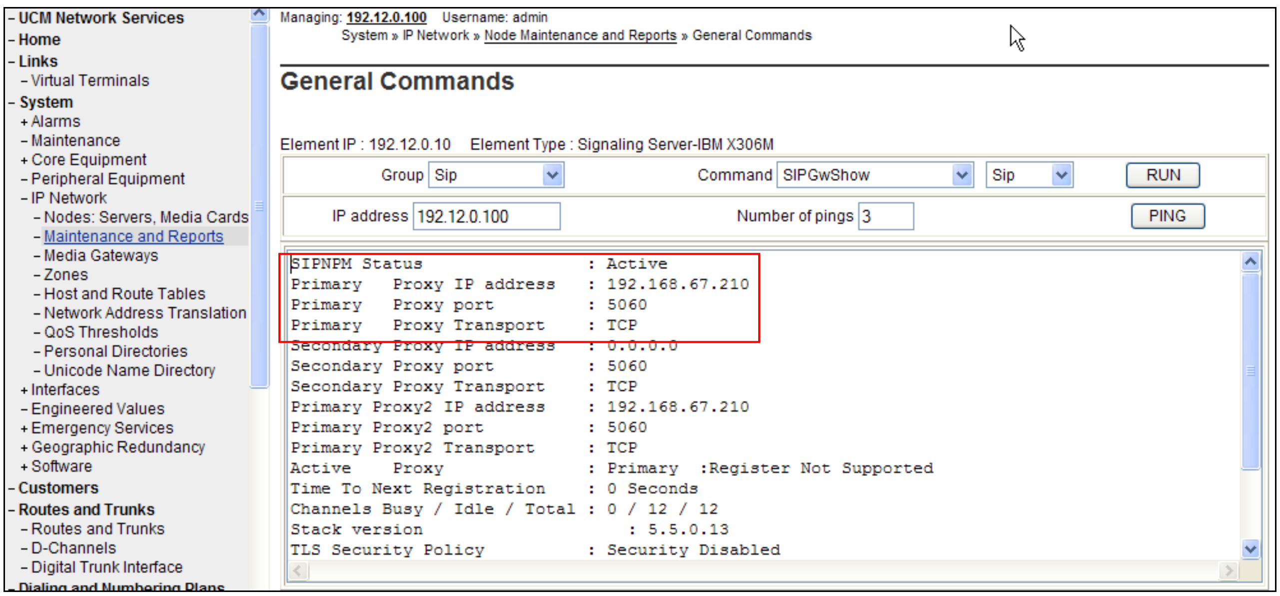



**Step 2** - In the resultant screen on the right, click the **Gen CMD** button. The **General Commands** page is displayed as shown below.

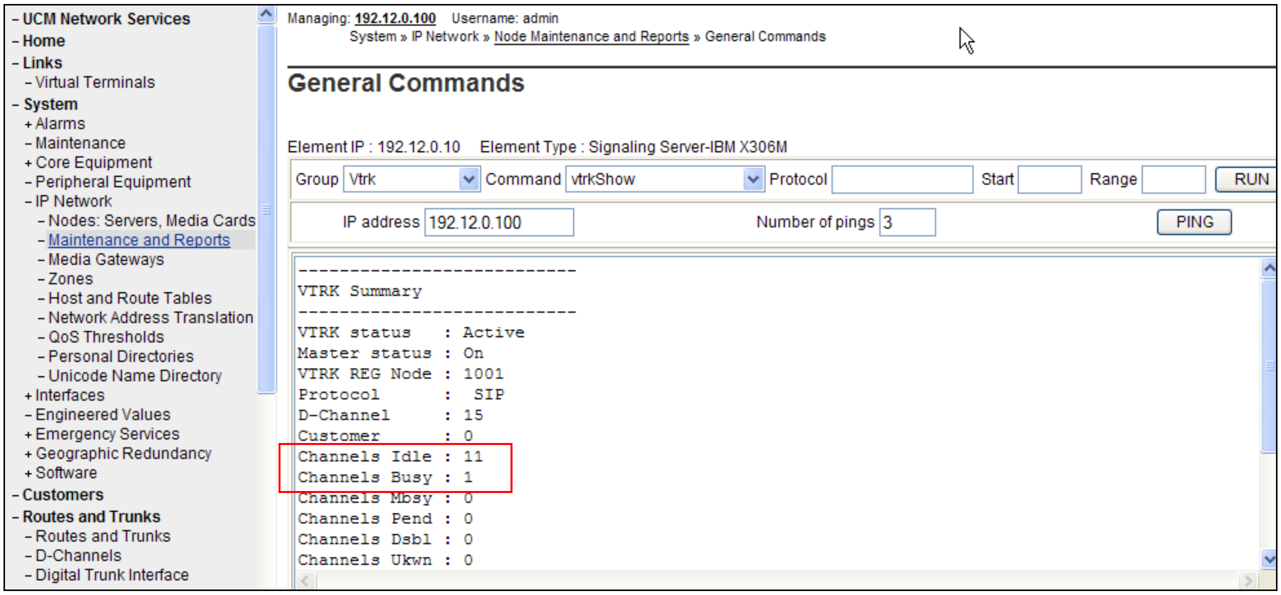



A variety of commands are available by selecting an appropriate **Group** and **Command** from the drop-down menus, and selecting **Run**.

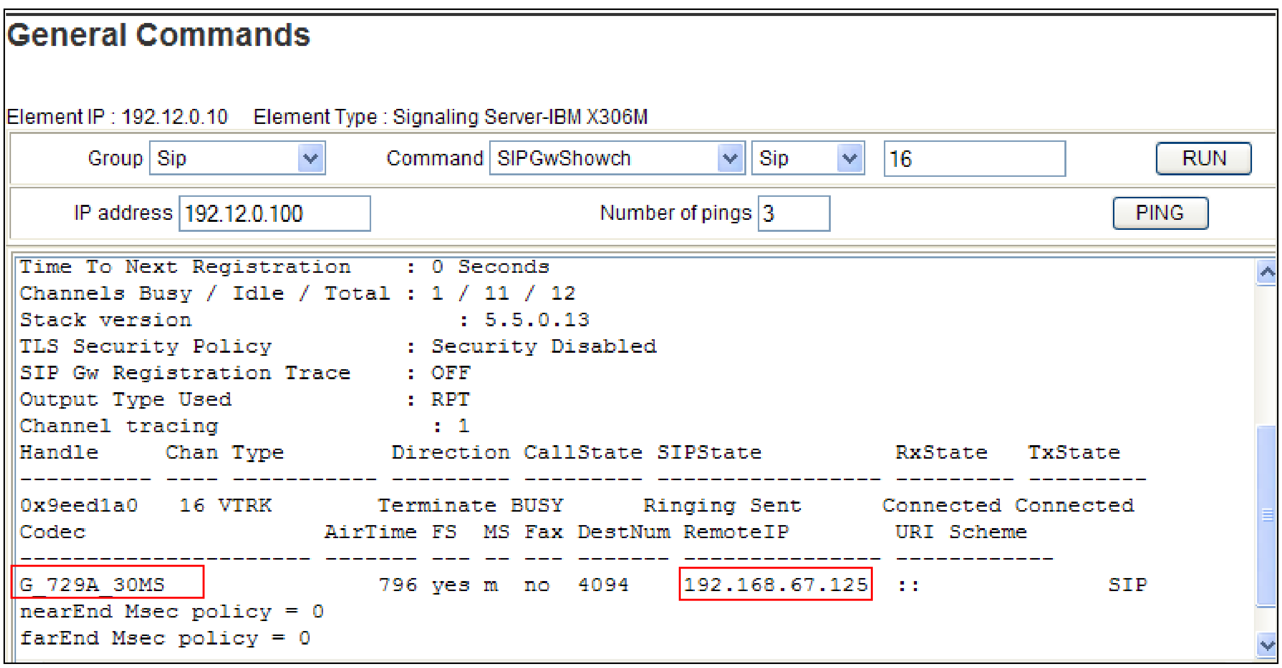
**Step 3** - To check the status of the SIP Gateway to Session Manager in the sample configuration, select "**Sip**" from the **Group** menu and "**SIPGwShow**" from the **Command** menu. Click **Run**. The example output below shows that the Session Manager (192.168.67.210, port 5060, TCP) has "SIPNPM Status" Active.




**Step 4** - As another example, the following screen shows the results of the "vtrkShow" **Command** from the "Vtrk" **Group**. The command was run with an active incoming PSTN call from the AT&T IP Flexible Reach service to an IP-UNIStim telephone. One channel is shown busy, and 11 idle.

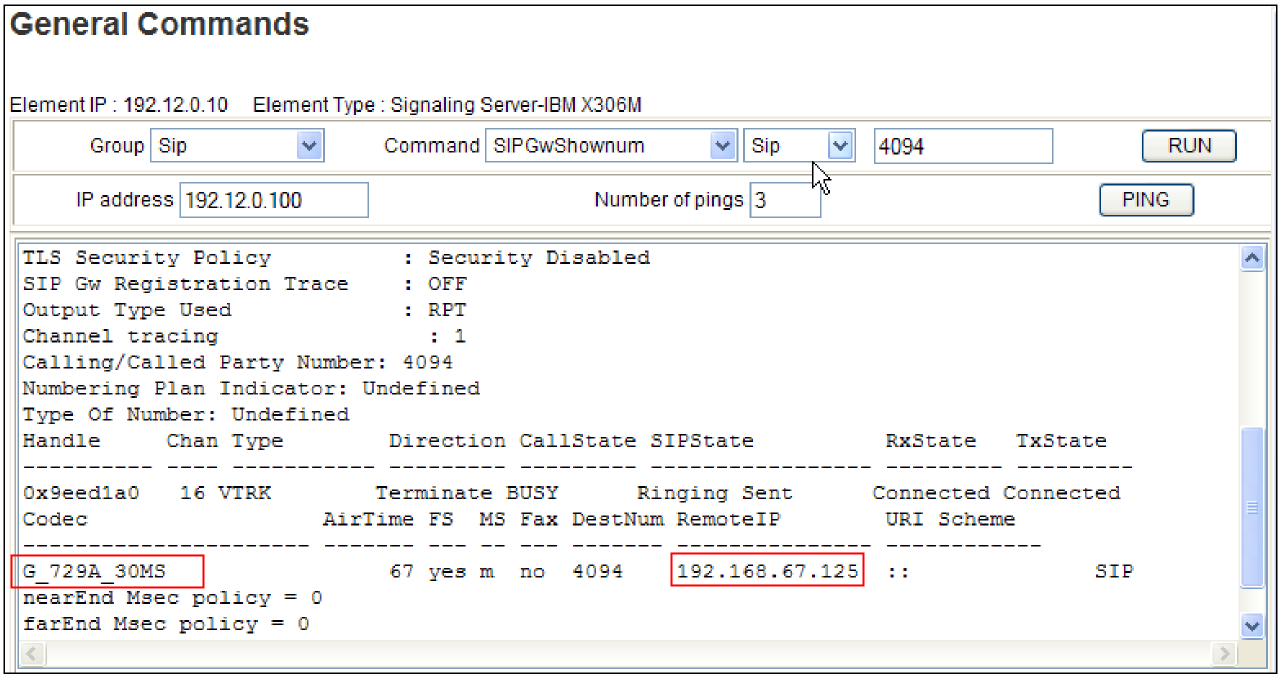



**Step 5** - The next screen capture shows the output of the **Command** "**SIPGWShowch**" in **Group** "**Sip**" for channel **16**[3], while an incoming call was active (using channel 16) from PSTN via the AT&T IP Flexible Reach service to an IP-UNIStim phone. In the output below, the scroll bar was used to scroll down to the area showing that the codec in use was "**G_729A_30MS**". Note that the Remote IP (**192.168.67.125**) is the IP Address of the inside private interface of the Avaya Aura® SBC.

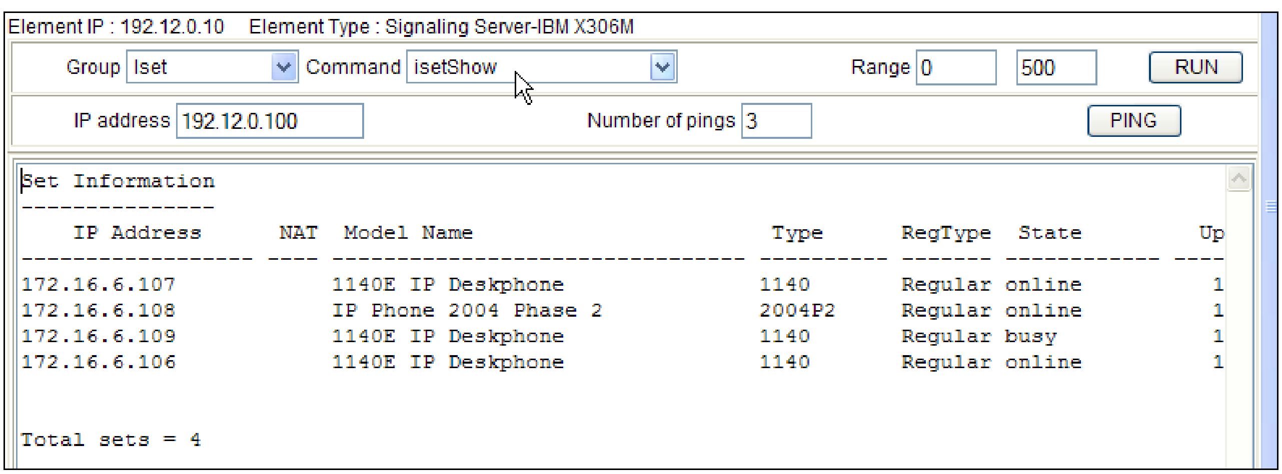



[3] Note – See **Section 5.2.2 Step 3** to determine the proper channel to display.

**Step 6** - The next screen capture shows an alternate way to view similar information, but in this case, by searching for calls involving a specific directory number. The screen shows the output of the **Command** "**SIPGWShownum**" in **Group** "**Sip**" where DN **4094** was specified. An incoming call was active from PSTN via the AT&T IP Flexible Reach service to the IP-UNIStim phone with DN 4094. In the output below, the scroll bar was used to scroll down to the area showing that the codec in use was "**G_729A_30MS**". Note that the Remote IP (**192.168.67.125**) is the IP Address of the inside private interface of the SBC.

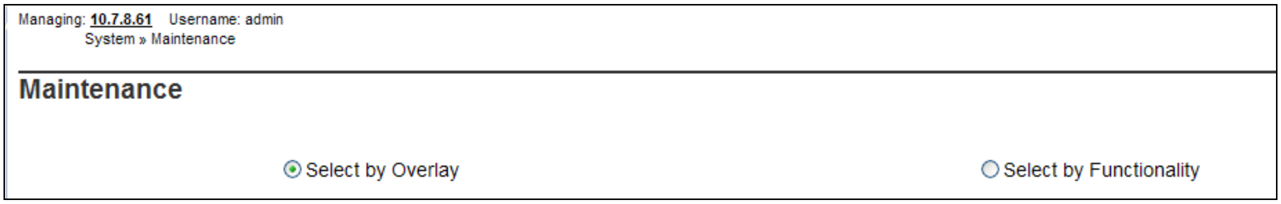



**Step 7** - The following screen shows a means to view IP UNIStim telephones. The screen shows the output of the **Command** "isetShow" in **Group** "Iset". At the time this screen was captured, the "4094 1140E IP Deskphone" UNIStim telephone was involved in an active call with PSTN via the AT&T IP Flexible Reach service.

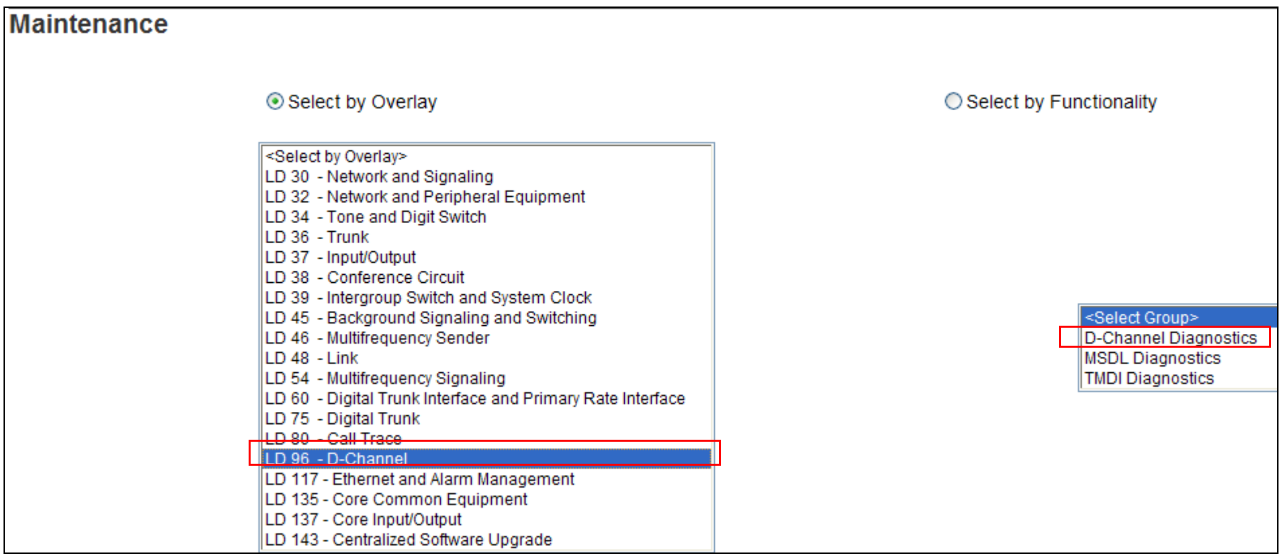


## 9.1.2 System Maintenance Commands

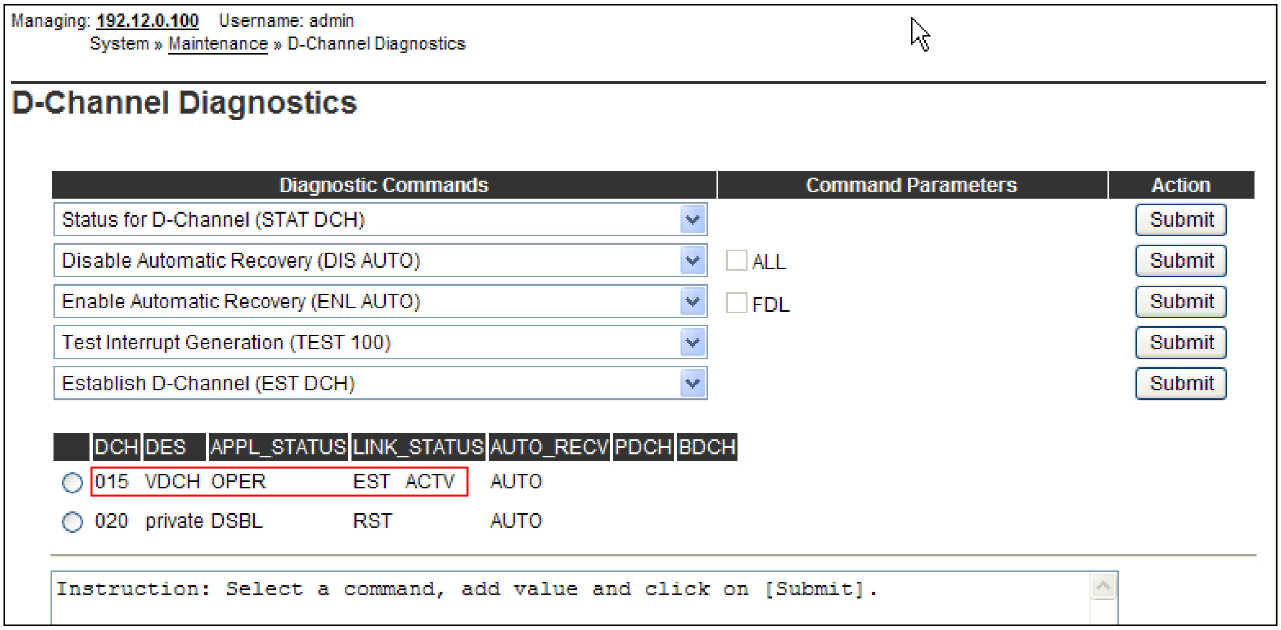
A variety of system maintenance commands are available by navigating to **System → Maintenance** using Element Manager. The user can navigate the maintenance commands using either the "**Select by Overlay**" approach or the "**Select by Functionality**" approach.




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-Channel number **15**, which is used in the sample configuration, is established "**EST**" and active "**ACTV**".


Managing: 192.12.0.100 Username: admin
System » Maintenance » D-Channel Diagnostics
D-Channel Diagnostics
Diagnostic Commands
Command Parameters
Action
Status for D-Channel (STAT DCH)
Disable Automatic Recovery (DIS AUTO)
Enable Automatic Recovery (ENL AUTO)
Test Interrupt Generation (TEST 100)
Establish D-Channel (EST DCH)
ALL
FDL
Submit
DCH DES APPL_STATUS LINK_STATUS AUTO_RECV PDCH BDCH
015 VDCH OPER EST ACTV AUTO
020 private DSBL RST AUTO
Instruction: Select a command, add value and click on [Submit].

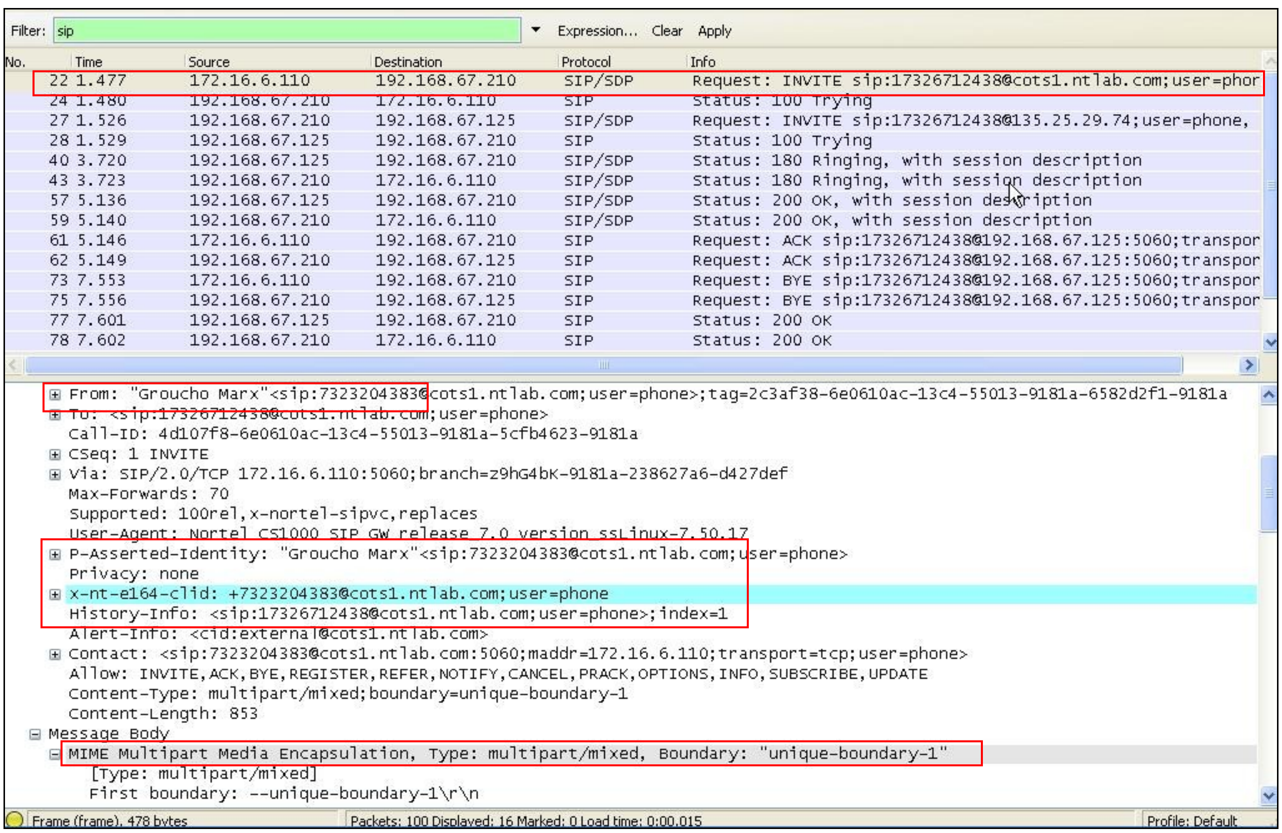
## 9.2. Wireshark Verifications

This section illustrates Wireshark traces for sample outbound and inbound calls using the reference configuration.

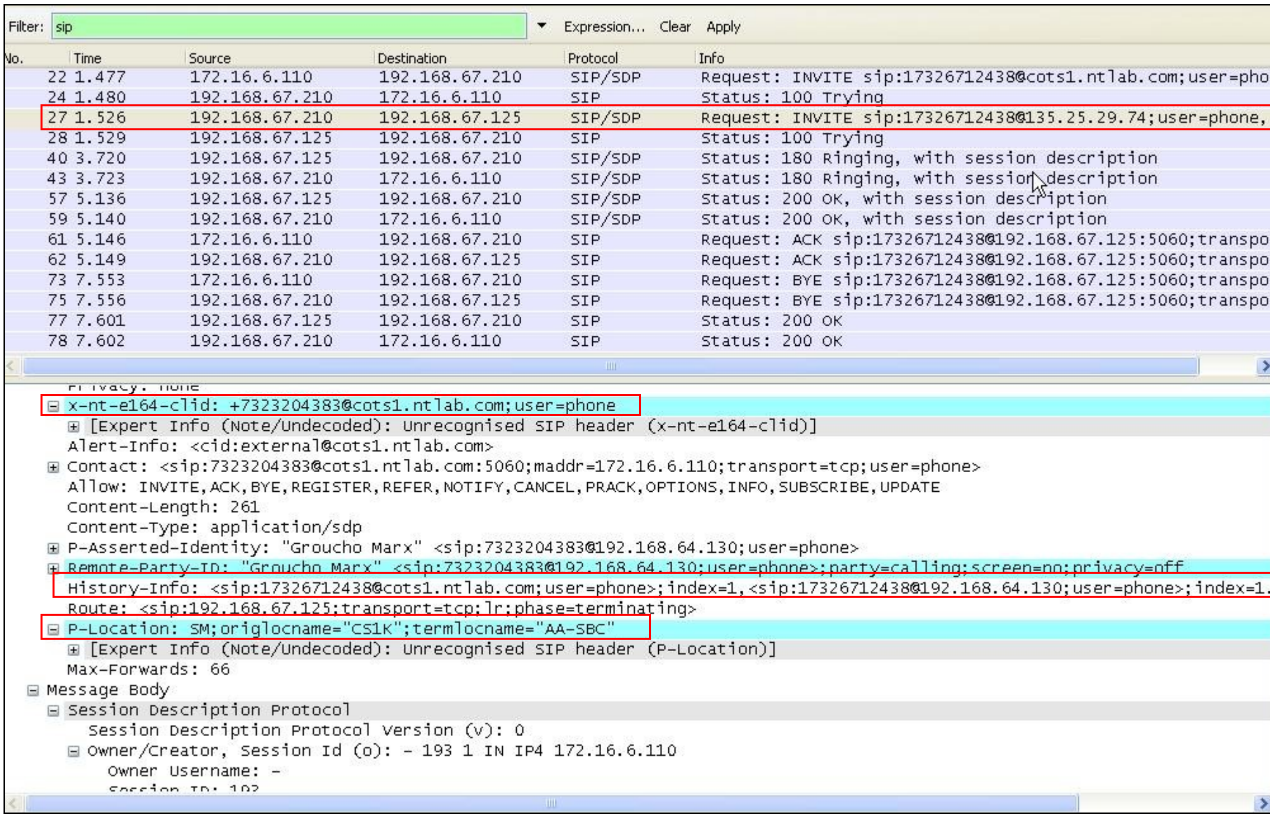
### 9.2.1 Example Outbound Call

This section illustrates an example outbound call from a CS1000E 1140E IP UNIStim user with Directory Number 4094 to PSTN.

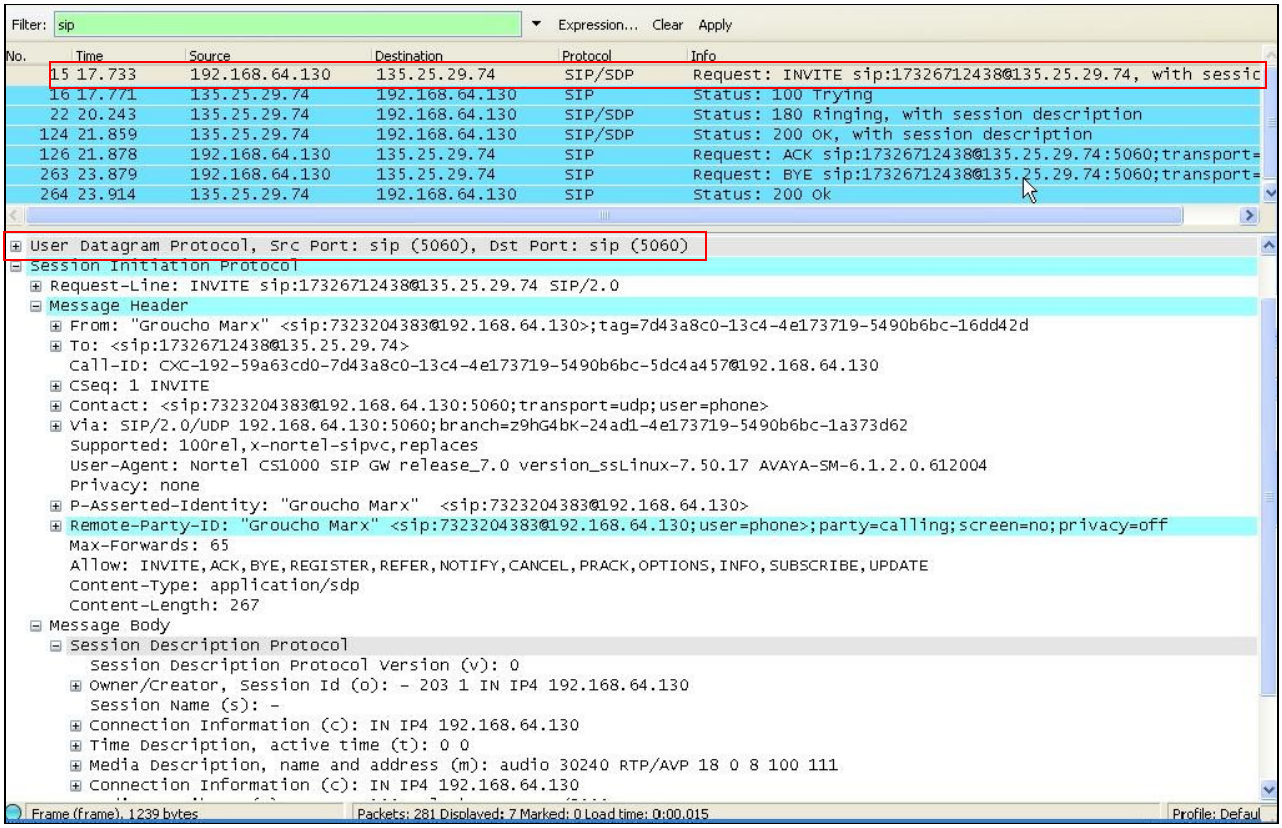
The following screen capture shows a Wireshark trace captured on the CPE private network, filtered on SIP messages. The INVITE message sent by the CS1000E is selected. As can be observed, in the sample configuration, the CS1000E sends the calling station's associated AT&T DID number (see **Section 5.8**) in SIP headers such as the From and P-Asserted-Identity headers. CS1000E proprietary headers such as "**x-nt-e164-clid**" can be observed, and such headers will be removed by the SBC. CS1000E **MIME** headers can be observed in the Message Body and will be removed by Session Manager. The **History-Info** header will be removed by the Avaya Aura® SBC.



The following screen capture shows the same Wireshark trace, however the Invite sent by Session Manager is selected. As can be observed from the selected section, the CS1000E proprietary header "**x-nt-e164-clid**" can still be observed, as is the **History-Info** header. Session Manager has inserted the **P-Location** header which will be removed by the Avaya Aura® SBC. However the CS1000E **MIME** header has been removed from the Message Body by Session Manager (see **Section 6.3.2**).

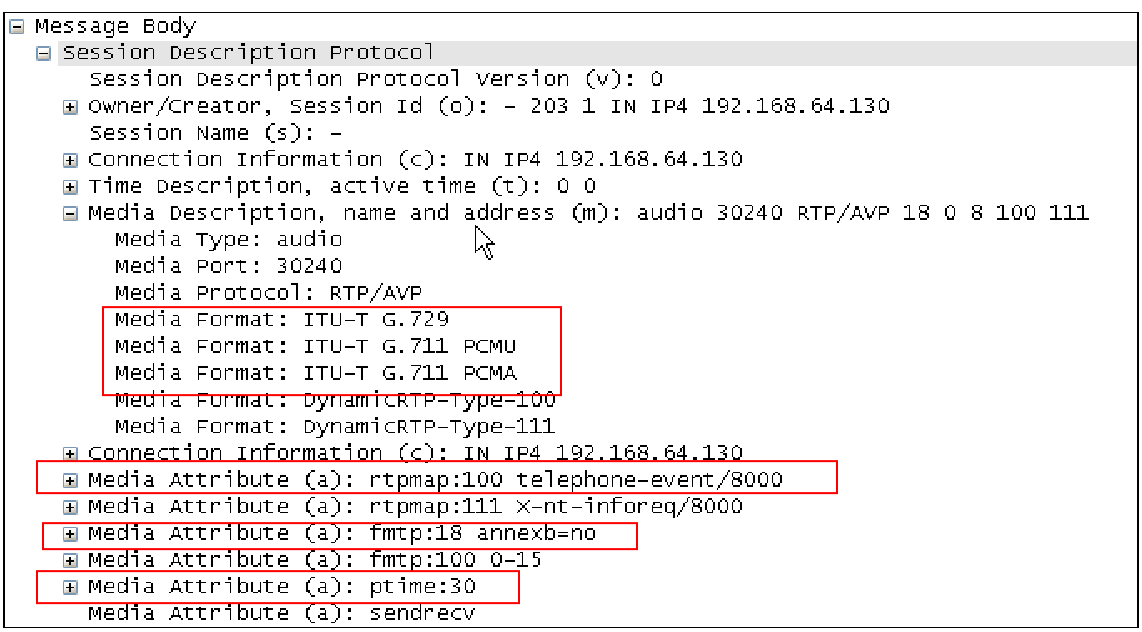


The following screen shows a Wireshark trace of the same outbound call, but taken at the public (outside) interface of the Avaya Aura® SBC. A portion of the INVITE sent to AT&T is shown (frame 15). The use of UDP and destination port 5060 can be observed (see **Sections 7.2.2** and **7.2.4.1**). Note that the CS1000E proprietary header "**x-nt-e164-clid**" and the **History-Info** header have been removed

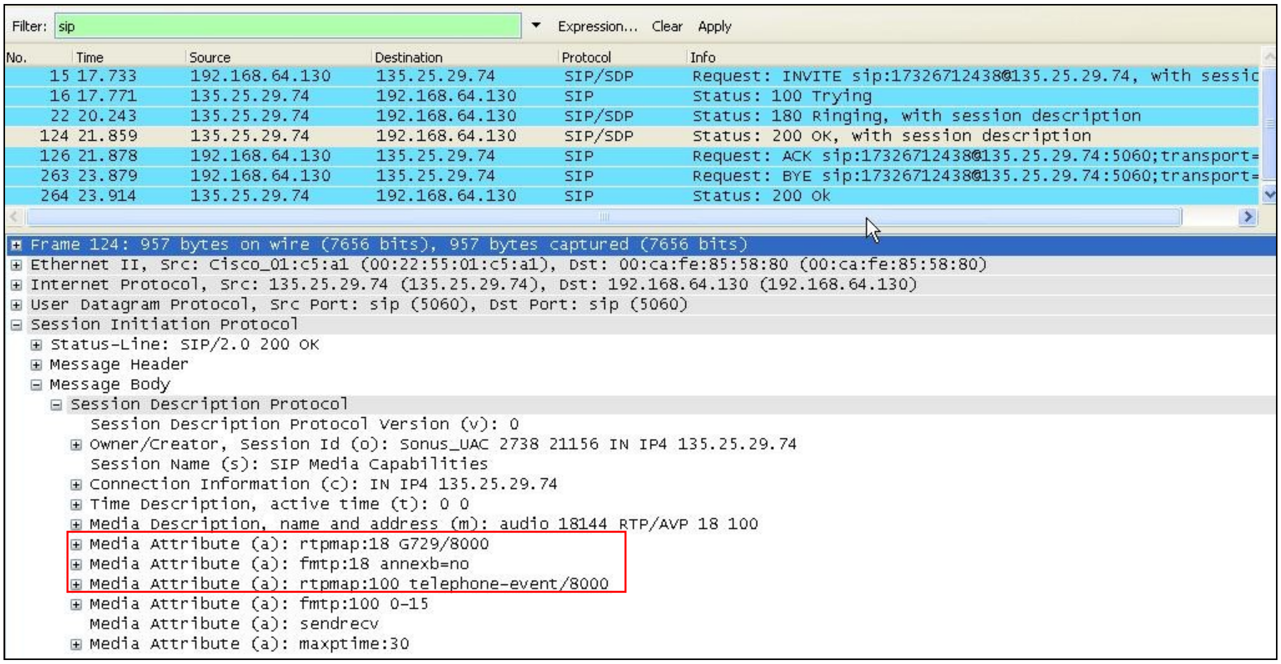



Scrolling down further in the Message Body shows the following:

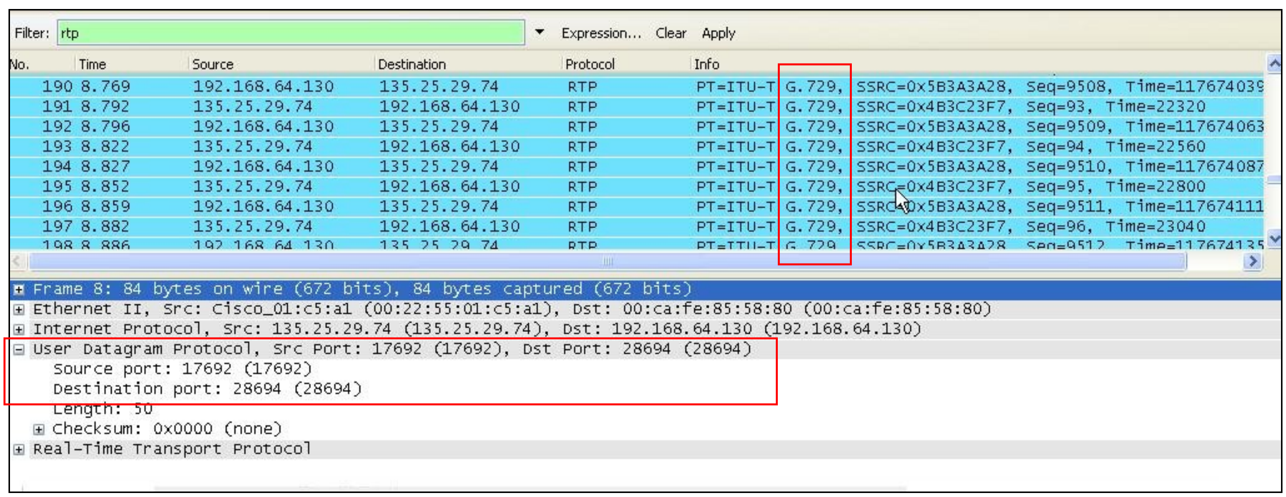
- The CS1000E offered G.729 and G.711 codecs as described in **Section 5.6**.
- RFC2833 DTMF Telephone Event 100 was set using the procedures described in **Section 5.10**.
- Annexb=no is specified, meaning G.729a is used.
- The ptime value of 30 provisioned in **Section 5.6** is also shown.



The AT&T IP Flexible Reach service responds with a 200OK that agrees on the use of G.729a, as well as also specifying RFC2833 DTMF Telephone Event 100.



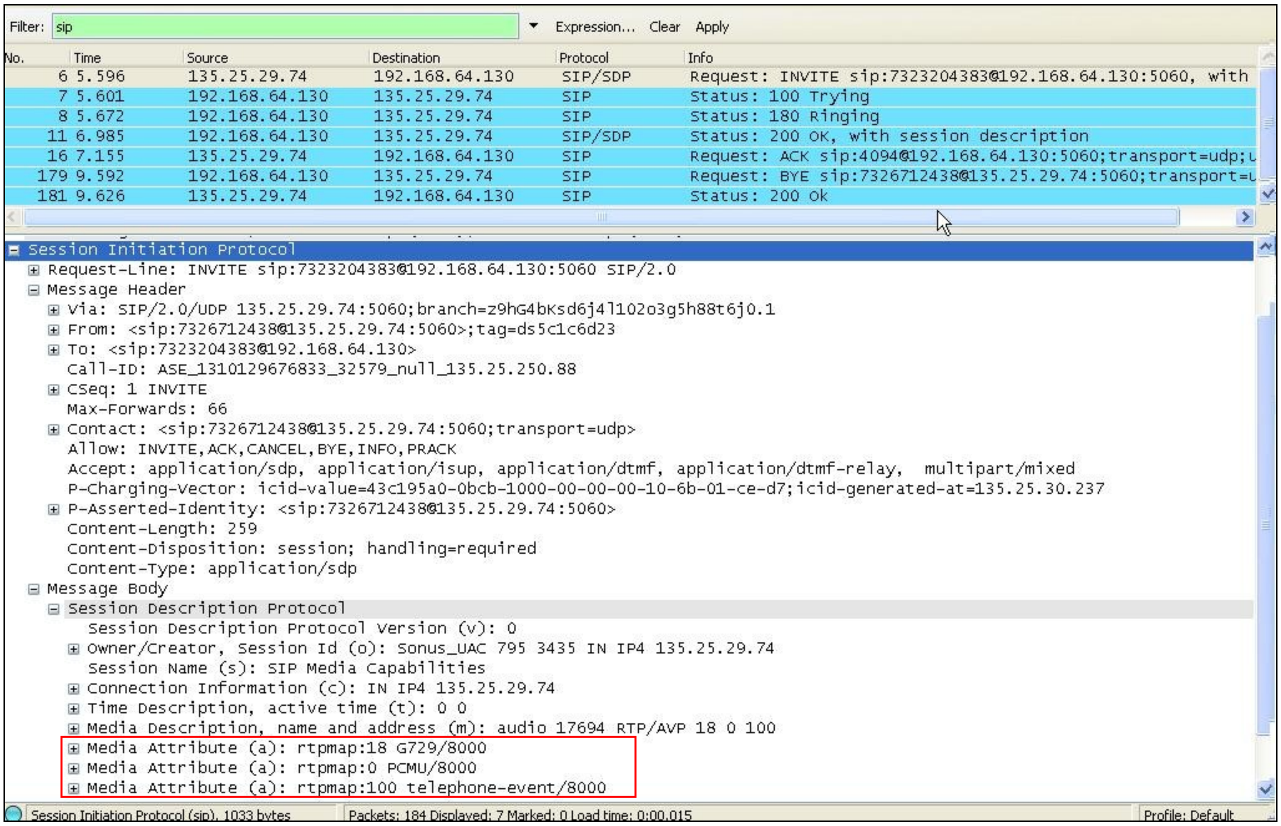

Changing the display filter to **rtp**, the media streams for this call are displayed. Note that the UDP ports used are within the range defined in **Section 7.2.3**. Also note that G.729 was the codec used.




## 9.2.2 Example Inbound Call

This section illustrates an inbound call from PSTN telephone 732-671-2438 to AT&T IP Flexible Reach DID 732-320-4383.

The following screen shows a Wireshark trace taken from the public interface of the Avaya Aura® SBC. Frame 6 shows an INVITE from AT&T, and is expanded to illustrate the contents of the message header and message body. Note that AT&T sends the calling party number 7326712438 in the From, Contact, and PAI headers. The Request-URI and To header both contain the dialed AT&T DID 7323204383. In the message body, note that the AT&T SDP offer lists G.729A and G.711mu. RFC2833 DTMF Telephone Event 100 is also specified.



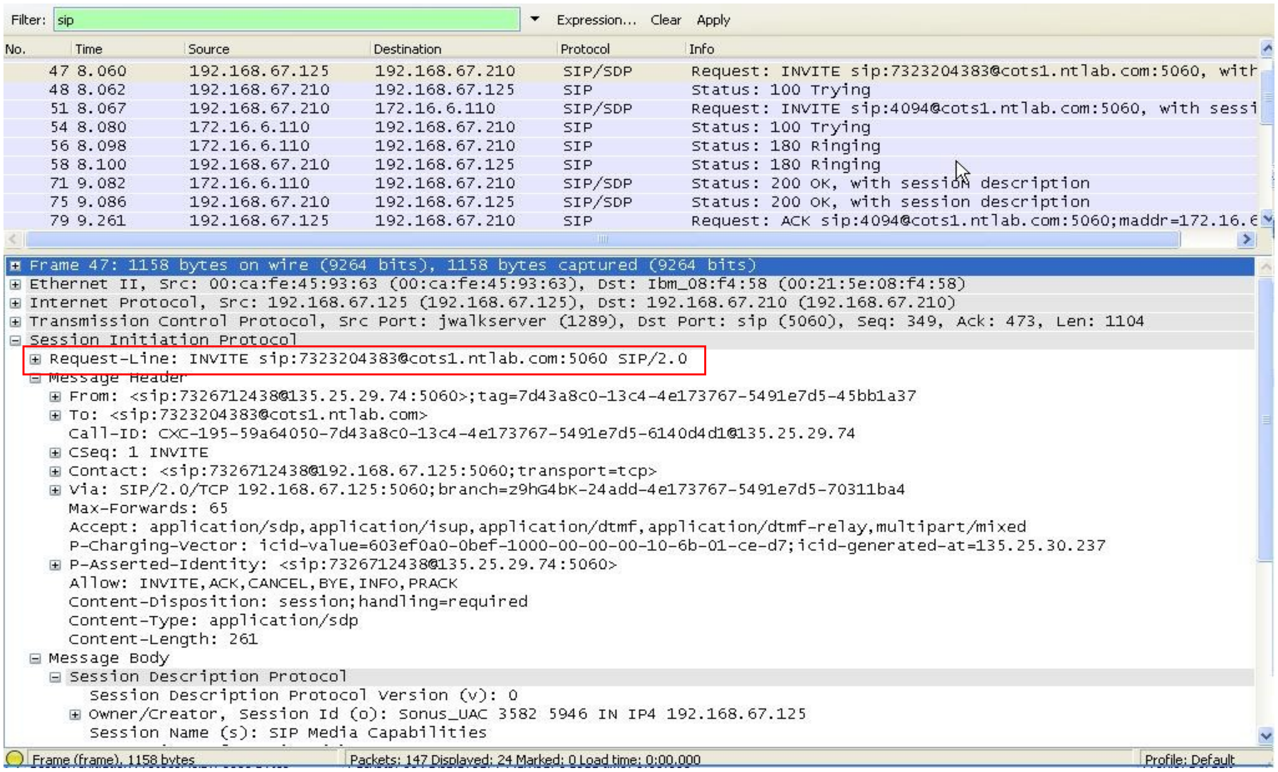

The following screen shows the 200 OK in frame 11 expanded to show the contents of the SDP Message Body from the CS1000E containing the following:

- G.729A with annexb=no
- RFC2833 DTMF Telephone Event 100
- Ptime=30

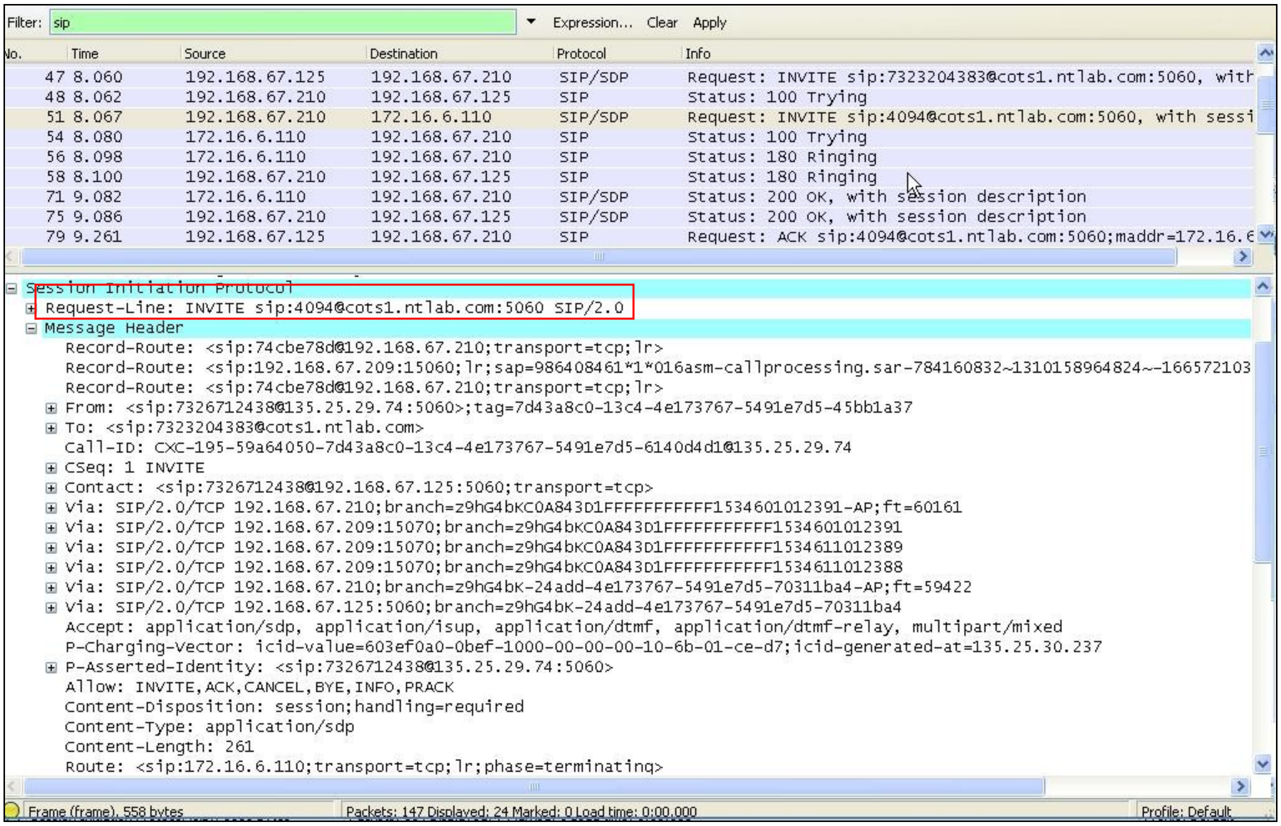
```
⊟ Message Body
  ⊟ Session Description Protocol
      Session Description Protocol Version (v): 0
    ⊞ Owner/Creator, Session Id (o): - 192 1 IN IP4 192.168.64.130
      Session Name (s): -
    ⊞ Connection Information (c): IN IP4 192.168.64.130
    ⊞ Time Description, active time (t): 0 0
    ⊟ Media Description, name and address (m): audio 28696 RTP/AVP 18 100 111
        Media Type: audio
        Media Port: 28696
        Media Protocol: RTP/AVP
        Media Format: ITU-T G.729
        Media Format: DynamicRTP-Type-100
        Media Format: DynamicRTP-Type-111
    ⊞ Connection Information (c): IN IP4 192.168.64.130
    ⊞ Media Attribute (a): rtpmap:100 telephone-event/8000
    ⊞ Media Attribute (a): rtpmap:111 X-nt-inforeq/8000
    ⊞ Media Attribute (a): ptime:30
    ⊞ Media Attribute (a): fmtp:18 annexb=no
    ⊞ Media Attribute (a): fmtp:100 0-15
      Media Attribute (a): sendrecv
```

Proceeding to the Wireshark from the inside of the SBC for this same call, Session Manager will modify the Request-URI from 732-320-4383 to CS1000E Directory Number 4094, an IP UNIStim 1140E telephone.

The following screen capture shows the INVITE message (frame 47) from the Avaya Aura® SBC inside interface to Session Manager. The message body contains the same called number and SDP offer information as shown in the previous screenshots.




The following screen capture shows the INVITE message (frame 51) from Session Manager to the CS1000E. The message body contains the same called number and SDP offer information as shown in the previous screenshots, however Session Manager has changed the Request-URI called number from 732-320-4383 to 4094 (see **Section 6.3.1**).




The following screen capture shows the same Wireshark trace but expands the 200 OK in frame 71 sent by the CS1000E when the user answers the call. The message body area is expanded to show the following:

- Media port within range 16384-32767
- G.729A with annexb=no
- RFC2833 DTMF Telephone Event 100
- Ptime=30

```
⊟ Message Body
  ⊟ Session Description Protocol
      Session Description Protocol Version (v): 0
    ⊞ Owner/Creator, Session Id (o): - 204 1 IN IP4 172.16.6.110
      Session Name (s): -
    ⊞ Connection Information (c): IN IP4 172.16.6.109
    ⊞ Time Description, active time (t): 0 0
    ⊟ Media Description, name and address (m): audio 16384 RTP/AVP 18 100 111
        Media Type: audio
        Media Port: 16384
        Media Protocol: RTP/AVP
        Media Format: ITU-T G.729
        Media Format: DynamicRTP-Type-100
        Media Format: DynamicRTP-Type-111
    ⊞ Connection Information (c): IN IP4 172.16.6.109
    ⊞ 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
```

## 9.3. System Manager and Session Manager Verification

This section contains verification steps that may be performed using System Manager for Session Manager verification.

### 9.3.1 Verify SIP Entity Link Status

Log in to System Manager. Expand **Elements** → **Session Manager** → **System Status** → **SIP Entity Monitoring**.

From the list of monitored entities, select an entity of interest, such as "AuraSBC". Under normal operating conditions, the **Link Status** should be "Up" as shown in the example screen below. The **Reason Code** column indicates that the Avaya Aura® SBC has responded to SIP OPTIONS from Session Manager with a SIP 404 message, which is sufficient for SIP Link Monitoring to consider the link up.

**SIP Entity, Entity Link Connection Status**

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

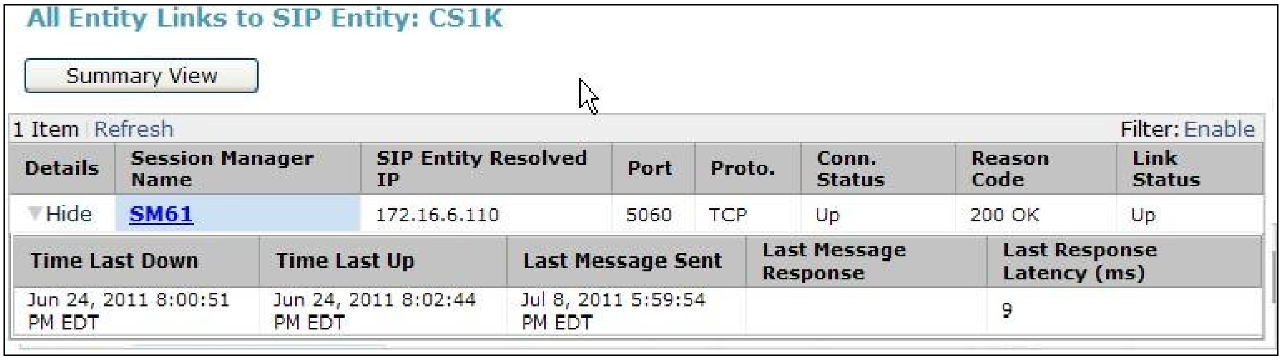
All Entity Links to SIP Entity: AA-SBC_to_AT&T

Summary View

1 Item Refresh Filter: Enable

| Details | Session Manager Name | SIP Entity Resolved IP | Port | Proto. | Conn. Status | Reason Code | Link Status |
|---|---|---|---|---|---|---|---|
| Show | SM61 | 192.168.67.125 | 5060 | TCP | Up | 404 Not found | Up |

Return to the list of monitored entities, and select another entity of interest, such as "CS1000-R75". Under normal operating conditions, the **Link Status** should be "Up" as shown in the example screen below. In this case, "Show" under Details was selected to view additional information.


All Entity Links to SIP Entity: CS1K

Summary View

1 Item | Refresh    Filter: Enable

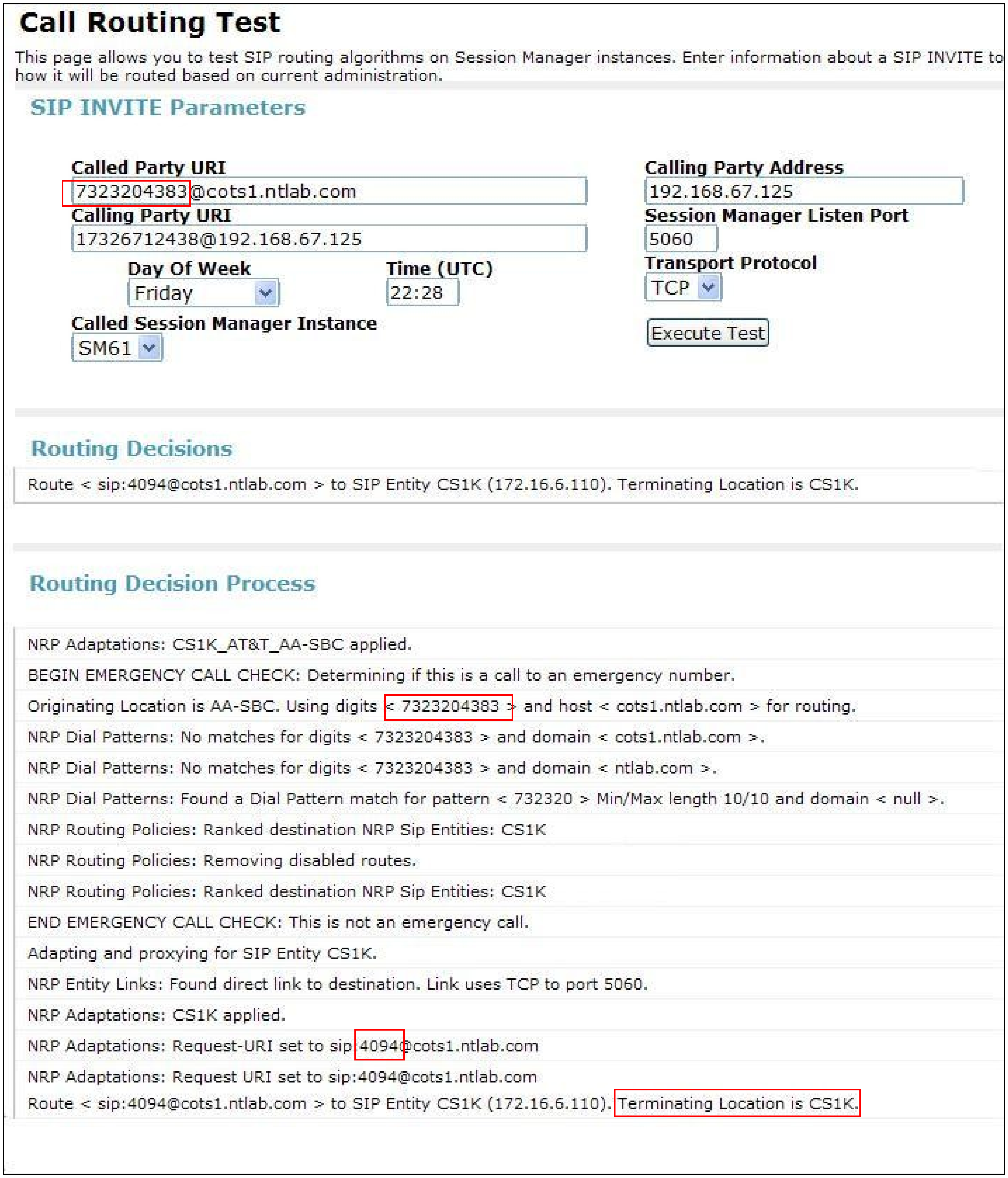
| Details | Session Manager Name | SIP Entity Resolved IP | Port | Proto. | Conn. Status | Reason Code | Link Status |
|---|---|---|---|---|---|---|---|
| ▼Hide | SM61 | 172.16.6.110 | 5060 | TCP | Up | 200 OK | Up |

| Time Last Down | Time Last Up | Last Message Sent | Last Message Response | Last Response Latency (ms) |
|---|---|---|---|---|
| Jun 24, 2011 8:00:51 PM EDT | Jun 24, 2011 8:02:44 PM EDT | Jul 8, 2011 5:59:54 PM EDT | | 9 |

### 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 CS1000K via AT&T. Note that the called number was AT&T DID 7323204383 and Session Manager converts this to CS1000E extension 4093 before routing the call to the CS1000E.



Call Routing Test
This page allows you to test SIP routing algorithms on Session Manager instances. Enter information about a SIP INVITE to
how it will be routed based on current administration.
SIP INVITE Parameters
Called Party URI
7323204383@cots1.ntlab.com
Calling Party URI
17326712438@192.168.67.125
Day Of Week
Friday
Time (UTC)
22:28
Called Session Manager Instance
SM61
Calling Party Address
192.168.67.125
Session Manager Listen Port
5060
Transport Protocol
TCP
Execute Test
Routing Decisions
Route < sip:4094@cots1.ntlab.com > to SIP Entity CS1K (172.16.6.110). Terminating Location is CS1K.
Routing Decision Process
NRP Adaptations: CS1K_AT&T_AA-SBC applied.
BEGIN EMERGENCY CALL CHECK: Determining if this is a call to an emergency number.
Originating Location is AA-SBC. Using digits < 7323204383 > and host < cots1.ntlab.com > for routing.
NRP Dial Patterns: No matches for digits < 7323204383 > and domain < cots1.ntlab.com >.
NRP Dial Patterns: No matches for digits < 7323204383 > and domain < ntlab.com >.
NRP Dial Patterns: Found a Dial Pattern match for pattern < 732320 > Min/Max length 10/10 and domain < null >.
NRP Routing Policies: Ranked destination NRP Sip Entities: CS1K
NRP Routing Policies: Removing disabled routes.
NRP Routing Policies: Ranked destination NRP Sip Entities: CS1K
END EMERGENCY CALL CHECK: This is not an emergency call.
Adapting and proxying for SIP Entity CS1K.
NRP Entity Links: Found direct link to destination. Link uses TCP to port 5060.
NRP Adaptations: CS1K applied.
NRP Adaptations: Request-URI set to sip:4094@cots1.ntlab.com
NRP Adaptations: Request URI set to sip:4094@cots1.ntlab.com
Route < sip:4094@cots1.ntlab.com > to SIP Entity CS1K (172.16.6.110). Terminating Location is CS1K.

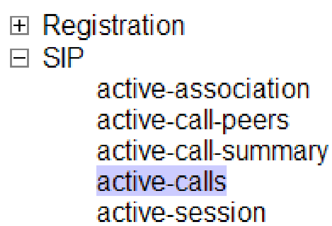
## 9.4. Avaya Aura® Session Border Controller Verification

This section contains verification steps that may be performed using the Avaya Aura® Session Border Controller.

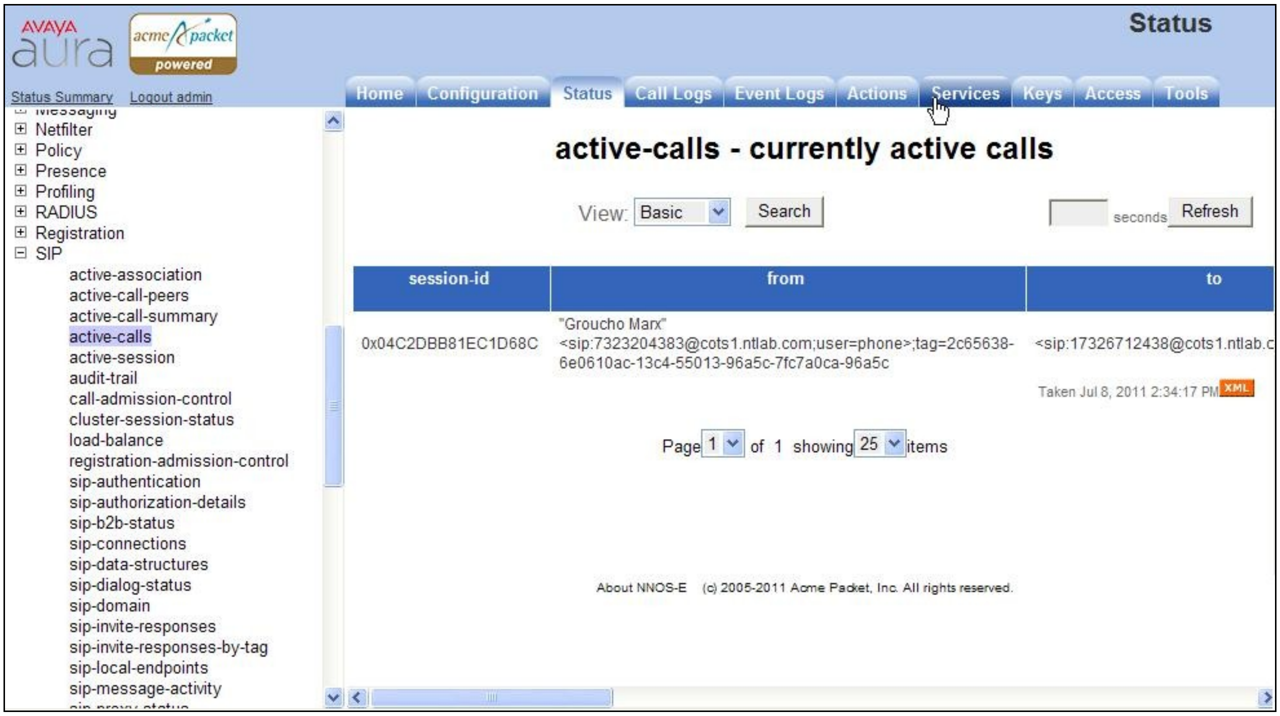
### 9.4.1 Status Tab

Avaya Aura® SBC status information is available via the **Status** tab.


For example, there is a SIP heading on the left menu that can be expanded as shown below.




In the example below, **active-calls** was selected from the left, revealing details about an active outbound call from a CS1000E 1140E Unistim station to PSTN. A scroll bar allows viewing of information about the active inbound call.

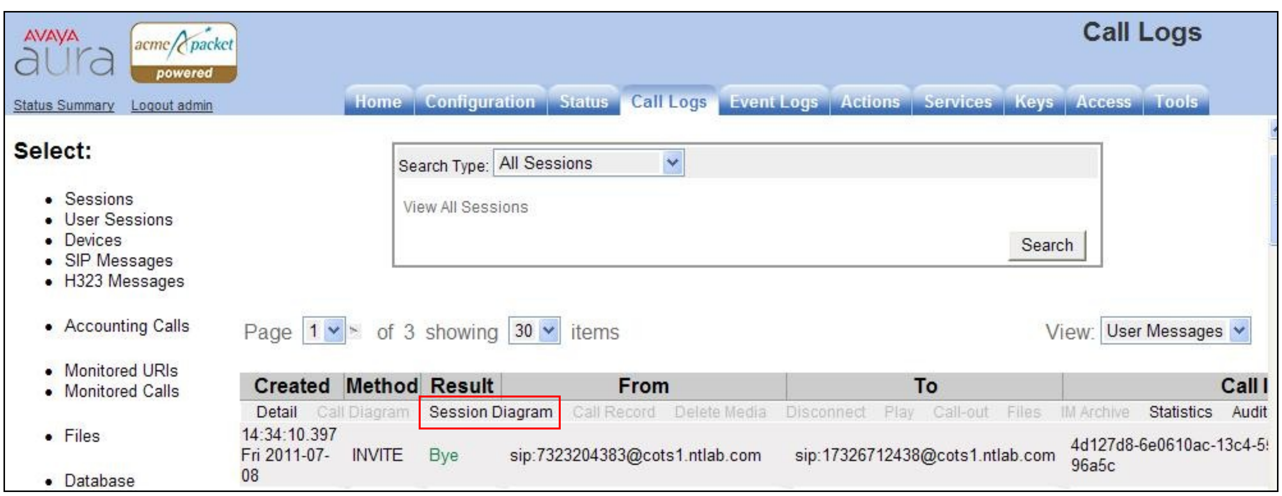



Additional information about the call is available by moving the bottom scroll bar to the right (not shown).

## 9.4.2 Call Logs

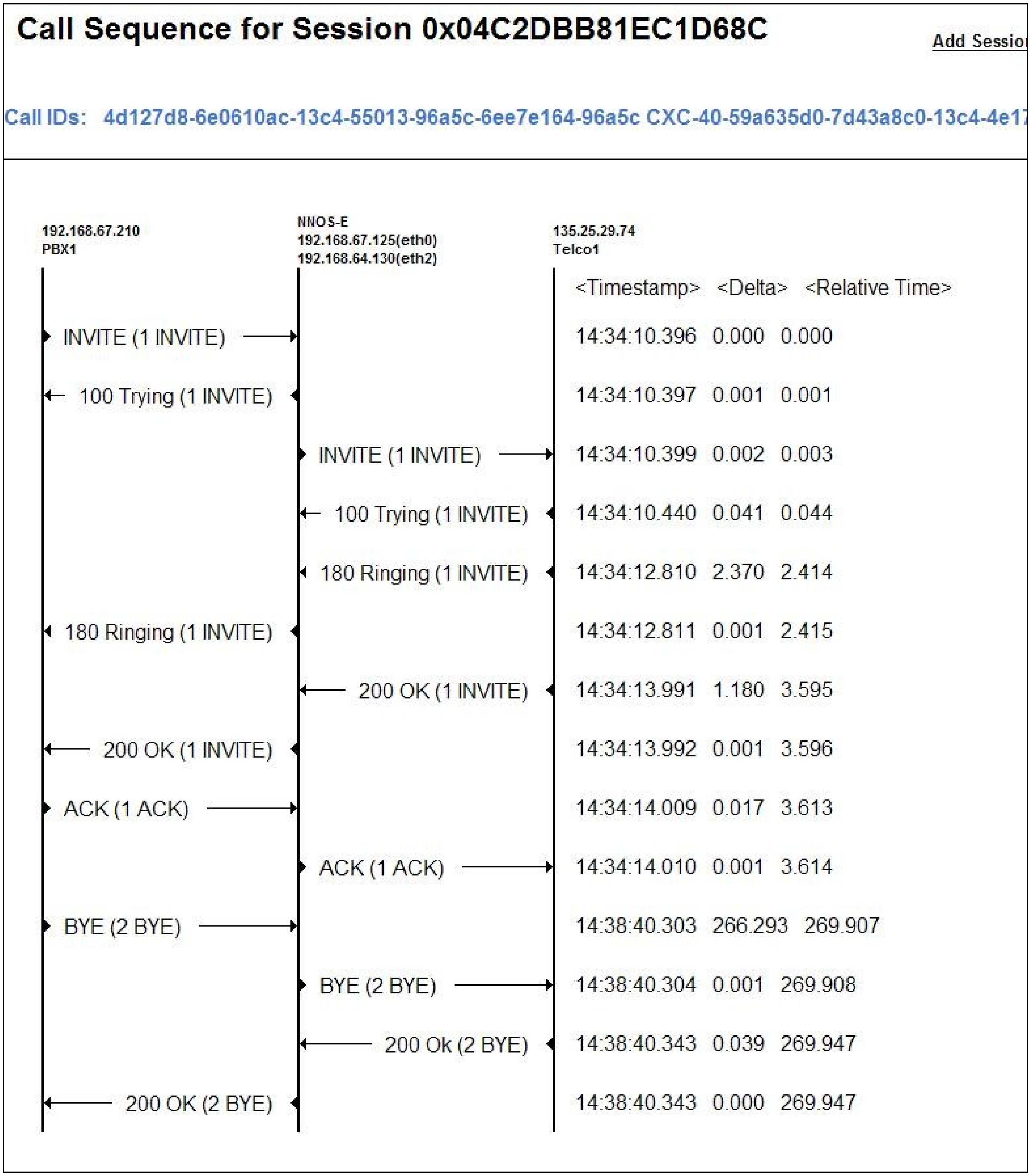
The **Call Logs** tab can provide useful diagnostic or troubleshooting information. In the following screen, the **SIP Messages** search capability can be observed.

The following screen shows a portion of the **Call Logs** tab selected after making an outbound call.

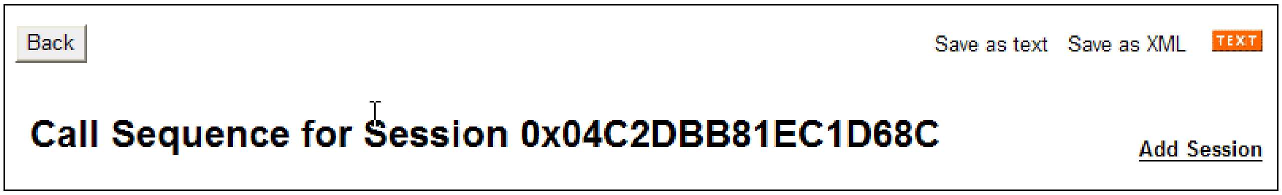



As shown below, select the **Session Diagram** link to view a ladder diagram for the session.

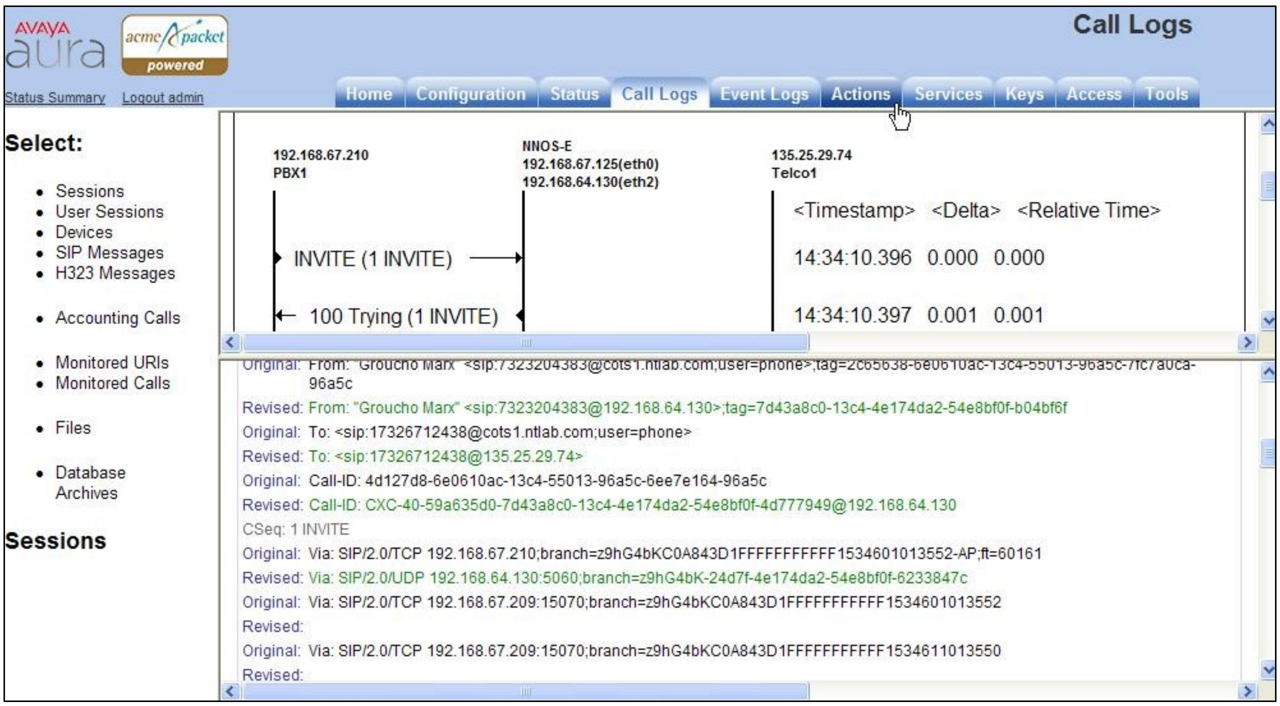
For example, the following screen shows a portion of the ladder diagram for the outbound call. Note that the activity for both the inside private and outside public side of the SBC can be seen.



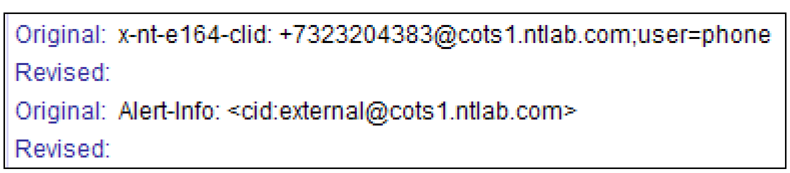

At the top right of the screen, the session may be saved as a text or XML file. If the session is saved as an XML file, using the **Save as XML** link, the xml file can be provided to support personnel that can open the session on another Avaya Aura® SBC for analysis.




The **Call Logs** tab also provides the capability to see modifications made to SIP headers by the SBC. Below the ladder diagram is another screen section. Using the same Session Diagram as shown above, Scrolling down to the INVITE message sent by the SBC to AT&T. The **More** and **See changes** links have been selected to expand the SIP message display and enable observation of the changes made by the SBC to the **Revised** message, as compared to the **Original** INVITE received from Session Manager.




Scrolling down further, the following screen shows that the SBC has deleted the “x-nt-e164-clid” and “Alert-Info” headers as defined in **Section 7.2.5**.




Scrolling down further the Avaya Aura® SBC removes the History-Info header as well as the P-Location Header as defined in **Section 7.2.5**.

| | |
|---|---|
| Original: | History-Info: <sip:17326712438@cots1.ntlab.com;user=phone>;index=1,<sip:17326712438@192.168.64.130;user=phone>;index=1.1 |
| Revised: | |
| Original: | P-Location: SM;origlocname="CS1K";termlocname="AA-SBC" |
| Revised: | |

# 10. Conclusion

As illustrated in these Application Notes, Avaya Communication Server 1000E Release 7.5, Avaya Aura® Session Manager Release 6.1, and the Avaya Aura® Session Border Controller Release 6.0.2 can be configured to interoperate successfully with AT&T IP Flexible Reach service via either AVPN or MIS-PNT transport. This solution allows Avaya Communication Server 1000E user access to the PSTN using an AT&T IP Flexible Reach service connection.

# 11. References

This section references documentation relevant to these Applications.

## 11.1. Avaya

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

**Avaya Aura™ Session Manager/System Manager**


[1] *Administering Avaya Aura™ Session Manager*, Doc ID 03-603324, Issue 4, Feb 2011 available at http://support.avaya.com/css/P8/documents/100082630

[2] *Installing and Configuring Avaya Aura™ Session Manager,* Doc ID 03-603473 Issue 2, November 2010 available at http://support.avaya.com/css/P8/documents/100089152

[3] *Maintaining and Troubleshooting Avaya Aura™ Session Manager,* Doc ID 03-603325, Issue 3.1, March 2011 available at http://support.avaya.com/css/P8/documents/100089154

[4] *Administering Avaya Aura™ System Manager*, Document Number 03-603324, June 2010 available at http://support.avaya.com/css/P8/documents/100089681


**Avaya Communication Server 1000E**


[5] *Communication Server 1000 Release 7.0 and Acme Packet Net-Net 6.2.0 Configuration Guide For Use with AT&T IP Flexible Reach,* Issue 1.1, 4/12/2011 available at: http://support.avaya.com/css/P8/documents/100129069

[6] IP Peer Networking Installation and Commissioning, Release 7.5, Document Number NN43001-313

[7] Unified Communications Management Common Services Fundamentals, Avaya Communication Server 1000E Release 7.5, Document Number NN43001-116

[8] Network Routing Service Fundamentals, Release 7.5, Document Number NN43001-130, Issue 03.02

[9] Co-resident Call Server and Signaling Server Fundamentals, Avaya Communication Server 1000E Release 7.5, Document Number NN43001-509

[10] Signaling Server and IP Line Fundamentals, Avaya Communication Server 1000E Release 7.5, Document Number NN43001-125

[11] Avaya Call Pilot® information can be found at:
http://support.avaya.com/css/Products/P0712

**Avaya Aura™ Session Border Controller**

[12] *Installing and Configuring Avaya Aura® Session Border Controller, Release 6.0.1, November 2010* available at:
http://support.avaya.com/css/P8/documents/100134970

[13] *Avaya Aura™ SBC System Administration Guide, V.6.0, 2010* available at:
http://support.avaya.com/css/P8/documents/100111137

[14] *Applications Notes for Avaya Aura™ Communication Manager 6.0, Avaya Aura™ Session Manager 6.0 and Avaya Aura™ Session Border Controller with AT&T IP Flexible Reach SIP Trunk Service – Issue 1.1, 2/18/2011* available at:
https://devconnect.avaya.com/public/download/dyn/CMSMAASBC60IPFR.pdf

Additional Avaya Application Notes on AT&T IP Flexible Reach service, tested via Avaya DevConnect, are available at the following link:

https://devconnect.avaya.com/dc/Public/WebListings/v2/CompanyWebListing.aspx?CompanyId=2262

## 11.2. AT&T IP Flexible Reach service.

Information regarding the AT&T IP Flexible Reach Service can be found at –

http://www.business.att.com/enterprise/Service/business-voip-enterprise/network-based-voip-enterprise/ip-flexible-reach-enterprise/

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.