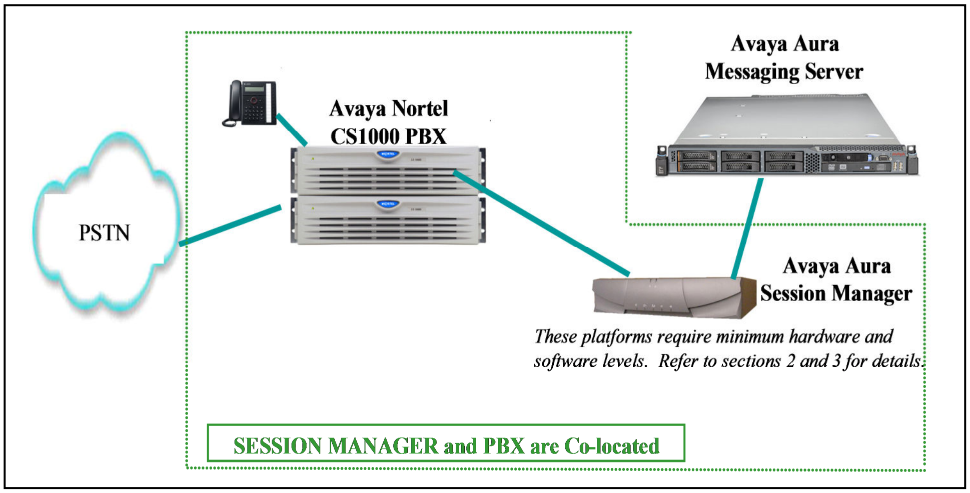
Avaya™

Aura Messaging

Configuration Note 88102 – Rev E (8/14)

Avaya CS1000

SIP Integration w/ Avaya Aura Session Manager



Note: Integrating **Aura Messaging with multiple PBXs** requires special consideration regarding Session Manager Administration to ensure call handling and MWI delivery. It is advisable to consult with your ATAC or Sales Engineer representative.

SIP Trunks allows the Avaya CS1000 PBX and the Avaya Aura Messaging Server to communicate over a LAN. For multiple tandem CS1K connections behind Session Manger, you MUST too use SIP trunk only to interconnect them.

Do not use PRI or H323 trunking.

Overview

This Configuration Note is intended for Avaya certified Aura Messaging technicians/engineers who are familiar with Aura Messaging procedures and terminology. It also assumes that you are Avaya certified or very familiar with the features and functionality of the Avaya PBXs supported in this Configuration Note and the SIP protocol.

Use this document in conjunction with *Aura Messaging Installation Guide* and the appropriate *Nortel PBX Guides* mentioned throughout this Configuration Note.

Please read the entire document before attempting any configuration.

1.0 METHOD OF INTEGRATION

The Session Initiation Protocol (SIP) integration provides connectivity with the Avaya PBX CS1000 over a Local Area Network (LAN). The connectivity between the Avaya Aura Messaging Server and the PBX is achieved over an IP-connected SIP trunk via the Aura Session Manager proxy. This integration passes call information and MWI using SIP packets.

Avaya Aura Messaging Server Requirements

p.- **Release Note**:

Should features of the integration not function optimally when integrated to a PBX or Aura Messaging that may be operating on an unsupported software release as defined Section 2.0 and 3.1, customers will need to upgrade their PBX and/or Aura Messaging to a supported software release.

2.0 AVAYA AURA MESSAGING SERVER REQUIREMENTS

- **Minimum releases required [1]:**
  - **Avaya Aura Messaging 6.x**

PBX hardware requirements

3.0 PBX HARDWARE REQUIREMENTS

Before performing the installation ensure the customer site has had an Avaya Network Assessment and the customer has implemented the recommendations.

- Avaya CS1000E CP+PM (Common Processor Pentium Mobile) Call Server 7.5 (with Software as detailed below in Section 3.1)

PBX/SESSION MANAGER software requirements

3.1 PBX SOFTWARE REQUIREMENTS

Minimum Software [1 (see pg 2)]:

- **Avaya CS1000E** updated to the current DEPLIST
- **Avaya CS1000E** with Release 7.5, Version 7.50.17

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3.2 SESSION MANAGER SOFTWARE/HARDWARE REQUIREMENTS

Minimum Supported Software and Hardware:

- Avaya Aura Session Manager 6.x

Hardware Required:

- Avaya S8xxx with SM100 card (*acts as gateway to SM*)
- Customer responsible for:
  - Monitor, Keyboard, and Mouse
  - Cat 5 Ethernet Cables
  - Blank DVDs for burning ISO images if needed

Please refer to *Installing and Administering Session Manager* for more details.

3.3 CONNECTIVITY

- Ethernet LAN connectivity – TCP/IP

3.4 CUSTOMER-PROVIDED EQUIPMENT

- Wiring/equipment necessary to support the physical LAN (CAT 5 minimum)

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Supported integration features

4.0 SUPPORTED INTEGRATION FEATURES

[✓] Items are supported

System Forward to Personal Greeting	
All Calls	[✓]
Ring/no answer	[✓]
Busy	[✓]
Busy/No Answer	[✓]
Station Forward to Personal Greeting	
All Calls	[✓]
Ring/no answer	[✓]
Busy	[✓]
Auto Attendant	[✓]
Call Me	[✓]
Direct Call	[✓]
External Call ID (ANI)	[✓]
Fax	[✓]
Find Me	[✓]
Internal Call ID	[✓]
Message Waiting Indication (MWI)	[✓]
Multiple Call Forward	[✓]
Multiple Greetings	[✓]
N+1	[✓]
Outcalling	[✓]
Queuing	[✓]
Return to Operator	[✓]

IMPORTANT: PBX options or features not described in this Configuration Note are not supported with this integration. To implement options/features not described in this document, please contact the Avaya Switch Integration product manager.

Classic Nortel "extended" proprietary features such as ESN, DSC and CDP topologies are not supported when connecting to Avaya PBX equipment such as CM or SM & must be disabled.

Multiple CS1K's are supported in a "flat" hierarchy with no overlapping extensions or mailboxes via SIP trunking between the CS1Ks.

DO NOT use H323 or PRI links between the tandem CS1Ks.

**** IMPORTANT ****

Encryption (TLS & SRTP) are currently not supported. Please disable "msec" (media security) or leave to "Best Effort" in which case AAM & CS1K will negotiate down to no security.

5.0 CONFIGURING THE AVAYA CS1000E

Note: This Configuration Note assumes basic configuration of telephones and SIP trunking to Session Manager has been completed.

For information on basic configuration please refer to *Communication Server 1000E Installation and Commissioning.* Release 7.5 Nortel Doc#NN43041-310.

PBX Configuration

The following tasks must be completed in the following order when programming the PBX to integrate. PBX programming is intended for certified PBX technicians/engineers.

*__Note__: Avaya uses the term "cover" while Nortel uses the term "forward."
For purposes of this document they are one in the same.

- Log in to CS1000E Element Manager
- Add a **Distant Steering Code** (DSC) for coverage and access to Aura Messaging
- Configure phones to cover* to the Aura Messaging 'pilot' extension
- Log in to the Network Routing Service (NRS)
- Add a route for the Aura Messaging 'pilot' extension

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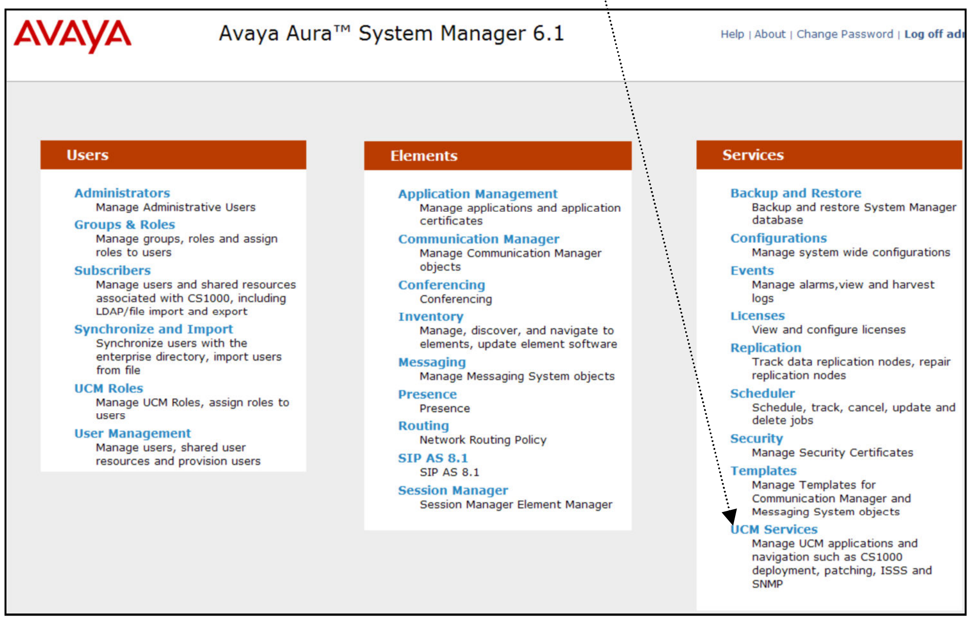
5.1 Configuring the Avaya CS1000E

- This configuration uses the Avaya Aura Unified Communication Management Server.
- Log in to the System Manager and choose UCM Services in the Services column on the right.



- This will bring you to the Avaya Unified Communications Management **Elements** page on the following page.

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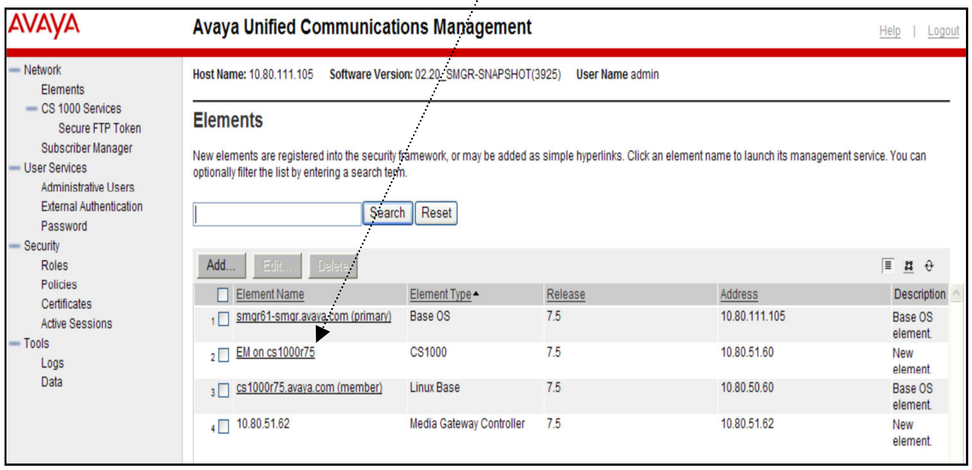
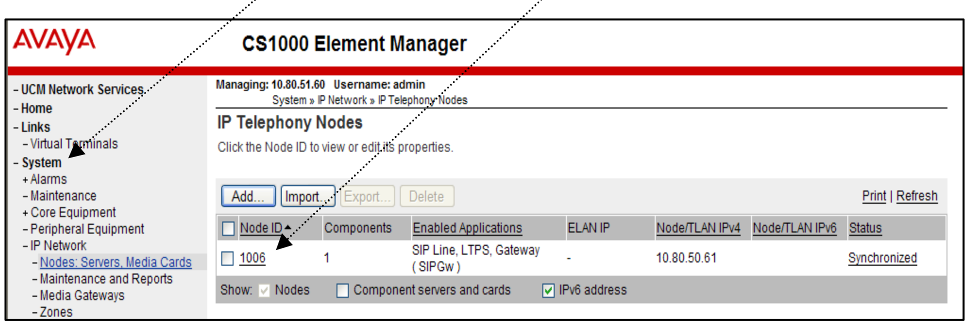
This section assumes the SIP trunk between Avaya Communication Server 1000E and Session Manager was already configured

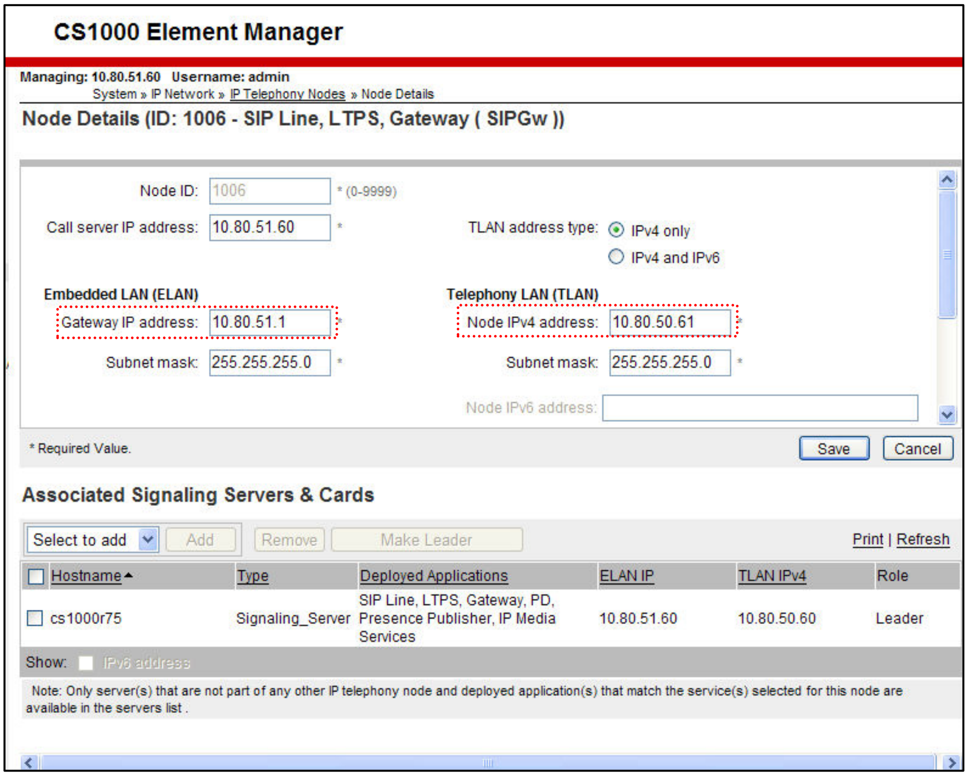
- On the screen below, click on the **Element Name** that corresponds to **CS1000** in the Element Type column.



- **Confirm Node and IP Addresses**
  - On the left side of the screen, expand **System,** and then under that **IP Network.**
  - Select **Nodes: Servers, Media Cards**.
  - The **IP Telephony Nodes** page is now displayed as shown below.
  - In the **Node ID** column click on your specific ID to view its details.
  - In our example configuration, **1006** is our Node ID.



- **Node Details**
  - The **Node Details** screen now appears as shown below.
  - Make a note of both the Embedded LAN **Call server IP address** and Telephony LAN **Node Ipv4 address** fields outlined below. These values will be used to configure other sections.



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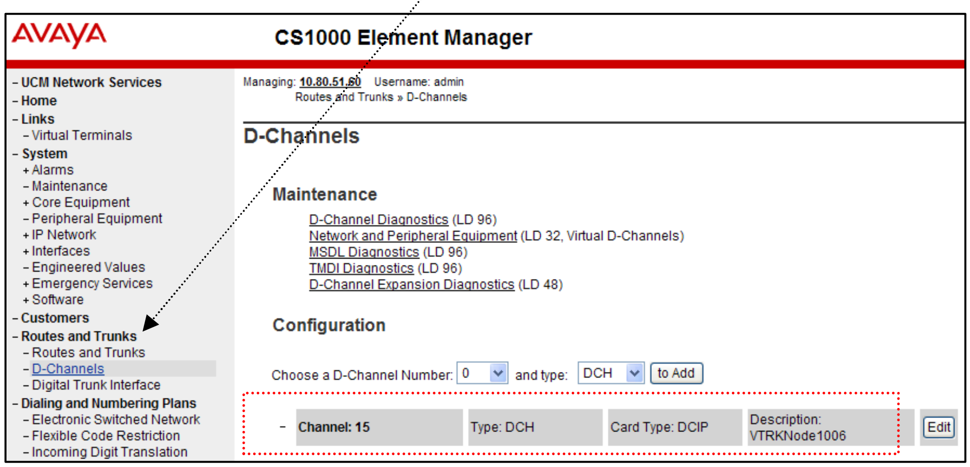
- **Confirm Virtual D-Channel Configuration**

The Avaya Communication Server 1000E Call Server uses a virtual D-channel and associated Route and Trunks to communicate with the Signaling Server. The following steps will guide you to ensure this administration was completed.

- On the left side scroll down until you see **Routes and Trunks.**
- Expand that section and select **D-Channels**.
- The screen below shows the D-channels administered on the sample configuration.
- In this configuration Channel 15 is our D-Channel, the Card Type is **DCIP.** This denotes the Channel is a *virtual* (IP) D-channel.



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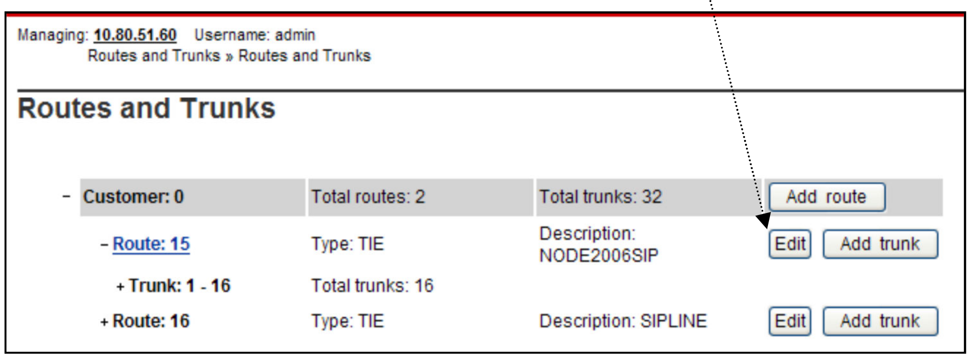
- **Confirm Routes and Trunks**
  - On the left side go to **Routes and Trunks** again.
  - When expanded there is a sub-menu that has an option with the same name **Routes and Trunks** (See screen on previous page).
  - Click on that sub-menu choice **Routes and Trunks.** The screen as shown below will appear.
  - This screen shows **Route 15** configured with the **Total Trunks** being **16**. This means the system can handle 16 concurrent calls.
  - To verify the configuration, select **Edit.**
  - The screen on the next page is now displayed.



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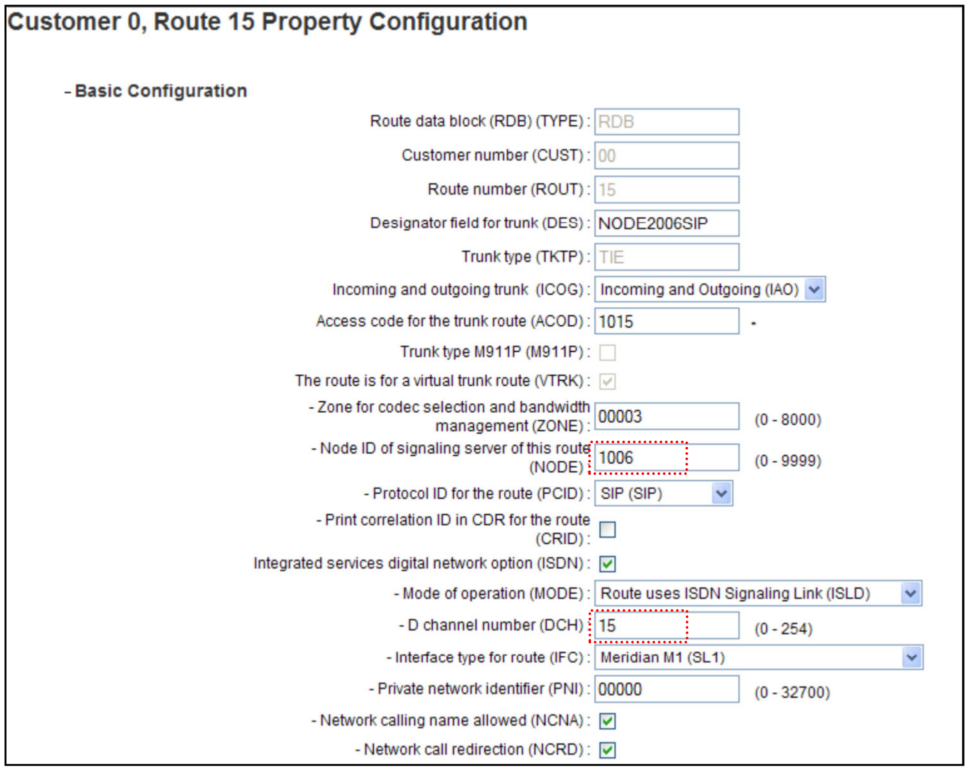
- **Details of the and Trunks**

Below are the Basic Configuration details of our virtual Route 15. Ensure the following are set.

  - Protocol ID for the route (PCID): **SIP (SIP)**
  - Node ID of signaling server of this route (NODE): **1006**
(This value matches the Node shown on the IP Telephony Screen on page 8)
  - D channel number (DCH): **15**
(This value matches the D-channel shown on the D-channels screen on page 10)


Customer 0, Route 15 Property Configuration

- Basic Configuration

Route data block (RDB) (TYPE) :	RDB	
Customer number (CUST) :	00	
Route number (ROUT) :	15	
Designator field for trunk (DES) :	NODE2006SIP	
Trunk type (TKTP) :	TIE	
Incoming and outgoing trunk (ICOG) :	Incoming and Outgoing (IAO)	
Access code for the trunk route (ACOD) :	1015	-
Trunk type M911P (M911P) :	☐	
The route is for a virtual trunk route (VTRK) :	☑	
- Zone for codec selection and bandwidth management (ZONE) :	00003	(0 - 8000)
- Node ID of signaling server of this route (NODE) :	1006	(0 - 9999)
- Protocol ID for the route (PCID) :	SIP (SIP)	
- Print correlation ID in CDR for the route (CRID) :	☐	
Integrated services digital network option (ISDN) :	☑	
- Mode of operation (MODE) :	Route uses ISDN Signaling Link (ISLD)	
- D channel number (DCH) :	15	(0 - 254)
- Interface type for route (IFC) :	Meridian M1 (SL1)	
- Private network identifier (PNI) :	00000	(0 - 32700)
- Network calling name allowed (NCNA) :	☑	
- Network call redirection (NCRD) :	☑	

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Important:

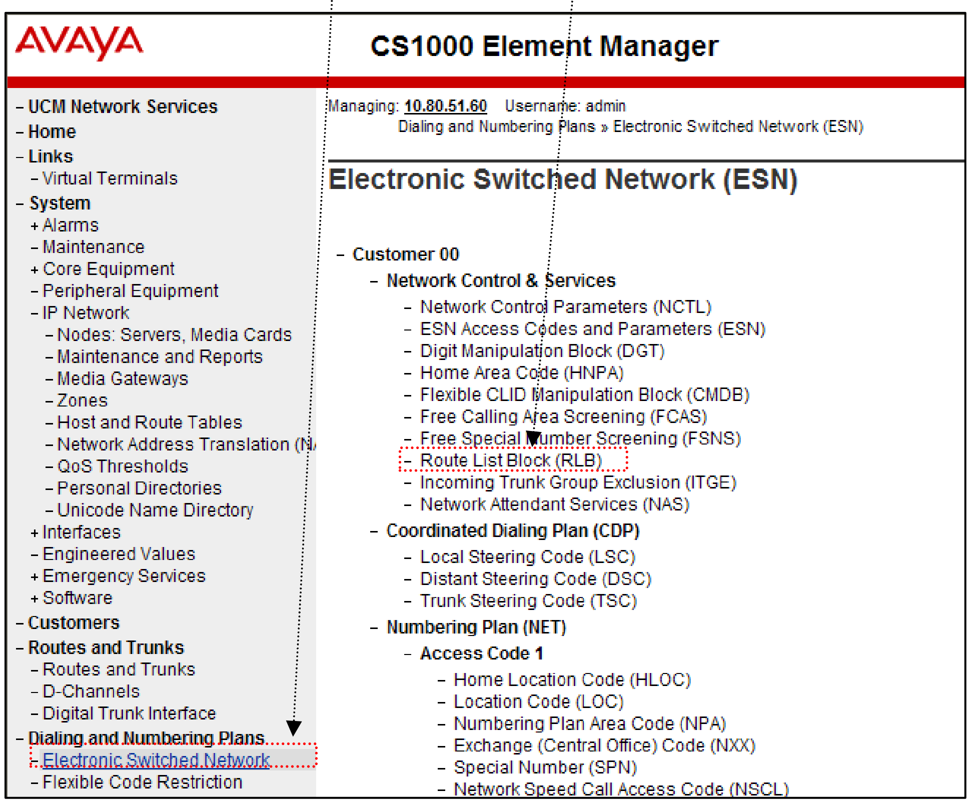
The Rules, Routing, and Distant Steering Codes (DSC) shown here are only examples.

Your Rules, Routing, and DSCs may be different for your customer network.

- **Route List Index and Distant Steering Code**

You will now configure the routing of calls to Avaya Aura® Messaging. You will first define the Route and then a Distant Steering Code.

- **Create Route List Index**
  - On the left side expand **Dialing and Numbering Plans.**
  - Select **Electronic Switched Network**.
  - Select **Route List Block (RLB)** on the right side of the **Electronic Switched Network (ESN)** screen shown below.



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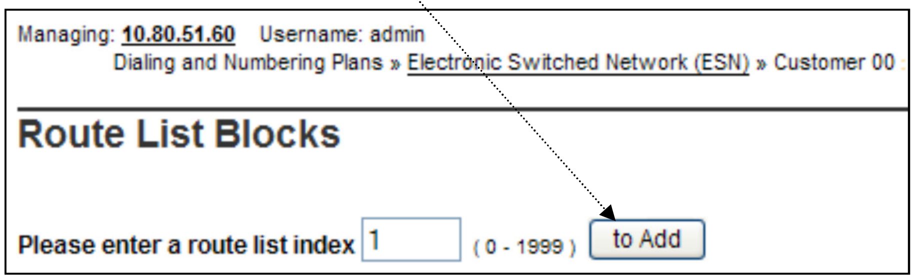
- The **Route List Blocks** screen is displayed.
    - In the **Please enter a route list index** field, enter an available route list index number
    - Click **to Add**



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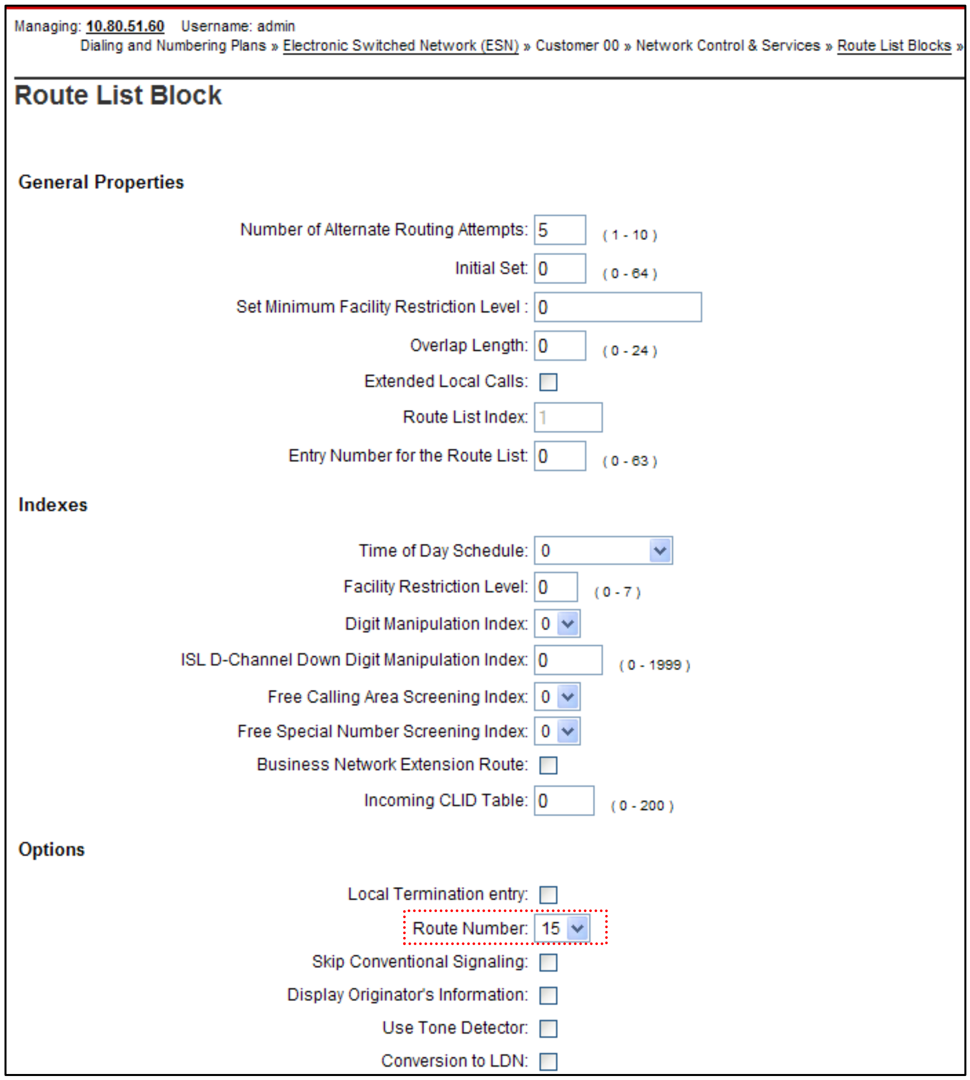
- You will now see the Route List Block Screen as shown below.
- Under the **Options** section
    - Select the **Routing Number**
      (This was the Route ID from Page 11 and 12)
    - Leave all remaining fields at their default values



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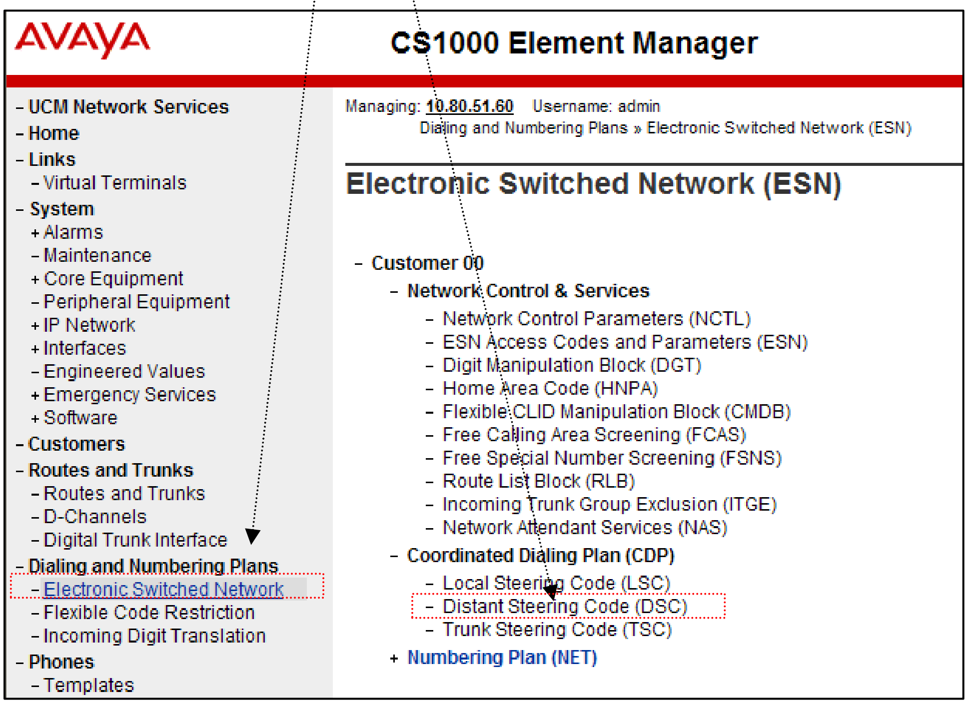
- **Add a Distant Steering Code (DSC)**
  - On the left of the of the ESN Screen (shown below), expand **Dialing and Numbering Plans**
  - Select **Electronic Switched Network**.
  - In the Coordinated Dialing Plan (CDP) on the right side, select **Distant Steering Code (DSC)**



  - You will now see a **Distant Steering Code List** screen, shown on the next page.

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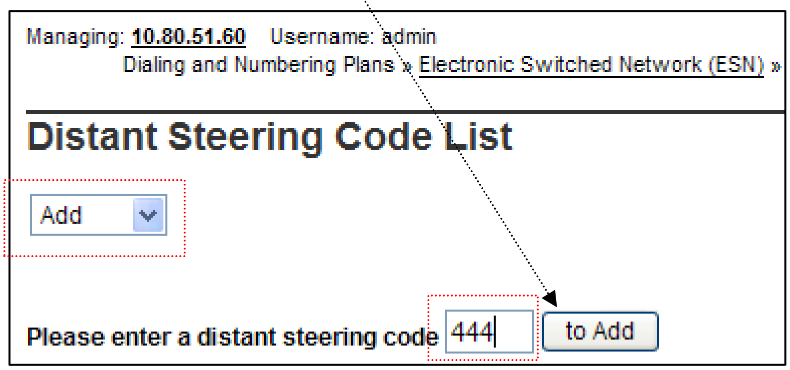
- **Distant Steering Code List**
  - Select **Add** from the drop-down menu
  - In the **Please enter a distant steering code** field, enter the dialed prefix for external calls that will be routed over the SIP trunk to Session Manager.

Note: In our sample configuration, our Distant Steering Code of **444** was used as the Avaya Aura® Messaging Pilot Number was **444-5000**, and the Auto Attendant number **444-5001**.

  - Click **to Add**

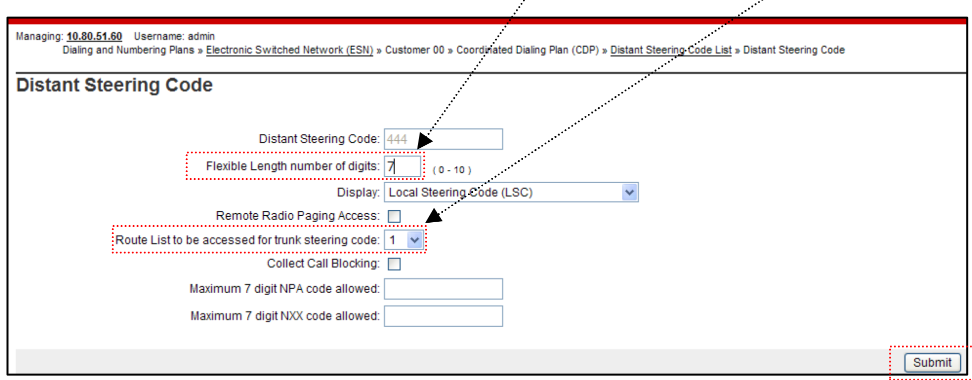


- **Enter the following for the fields in the screen below**:
  - Flexible Length number of digits: **7**

(This is the number of digits in dialed numbers. In our configuration we used 7 digit numbers)

  - Route List to be accessed for trunk steering code: **1**

(This is the Route List Index we created in the Route List Blocks on page 14)

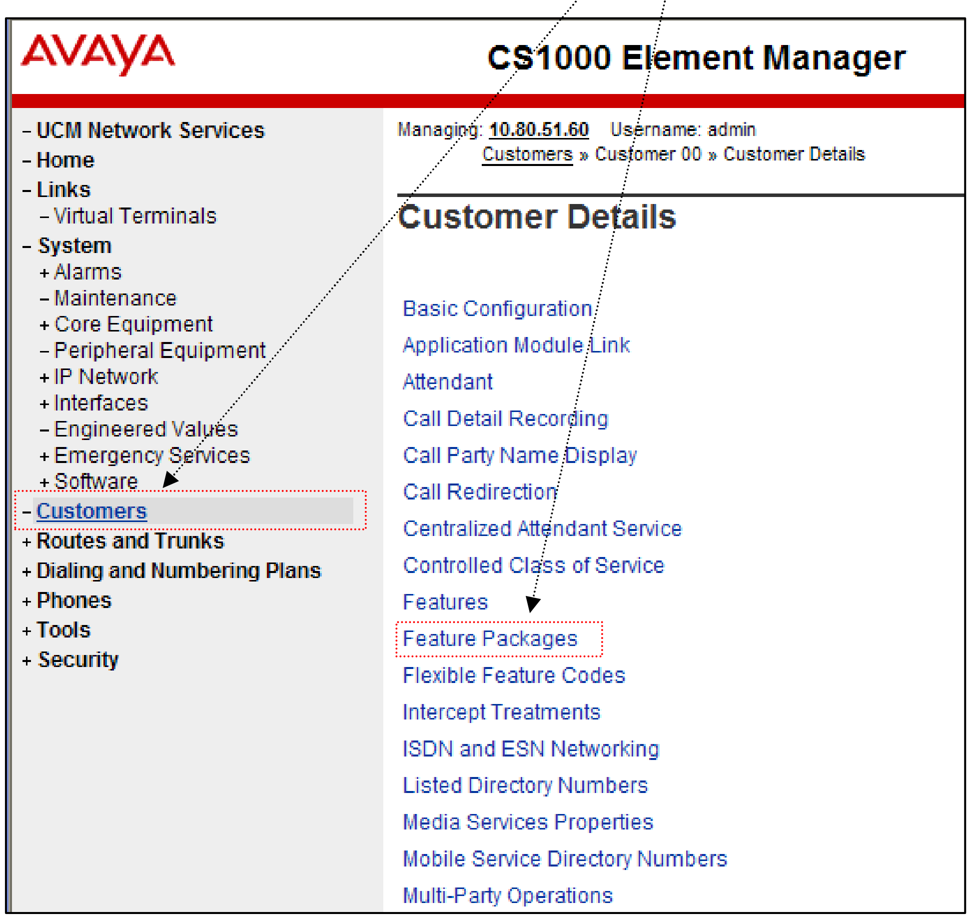


  - Click **Submit** to save

- **Private Network Identifier & MWI**

To activate MWI, notify messages are sent from Avaya Aura® Messaging to the Avaya CS1000E. To enable Avaya Communication Server 1000E to receive SIP Notify messages from Avaya Aura® Messaging you need to administer a Private Network Identifier for the system.

- On the left side of the screen (below) expand **Customers**.
  - Select the ***customer*** (not shown in the screen below)
  - On the right side select **Feature Packages**



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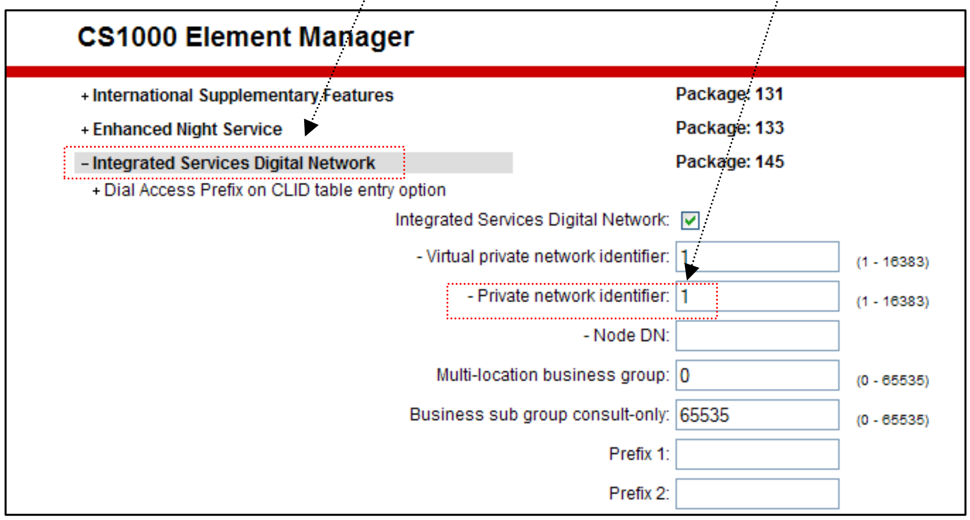
- You will now see the **Feature Packages** page (not shown)
- Expand **Integrated Services Digital Network**
- In the **Private network identifier** field, enter **1***

*__Note__: A Private Network Identifier of **1** was used in our configuration. Your Private Network Identifier may be different.



- Click **Save**

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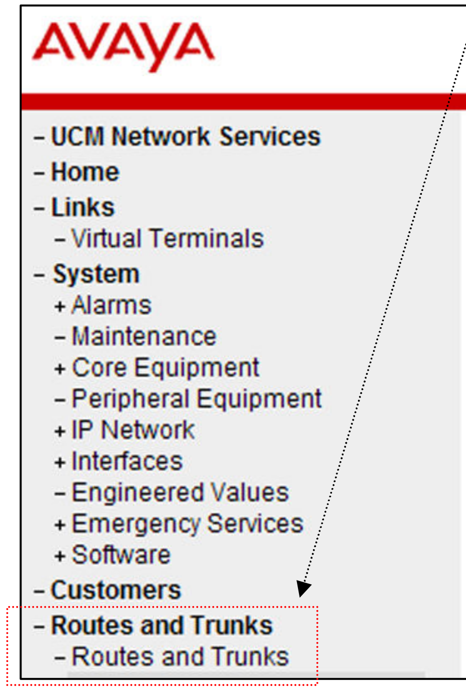
- **Routes and Trunks**
  - On the left side expand **Routes and Trunks**
  - Click on the sub-group **Routes and Trunks**



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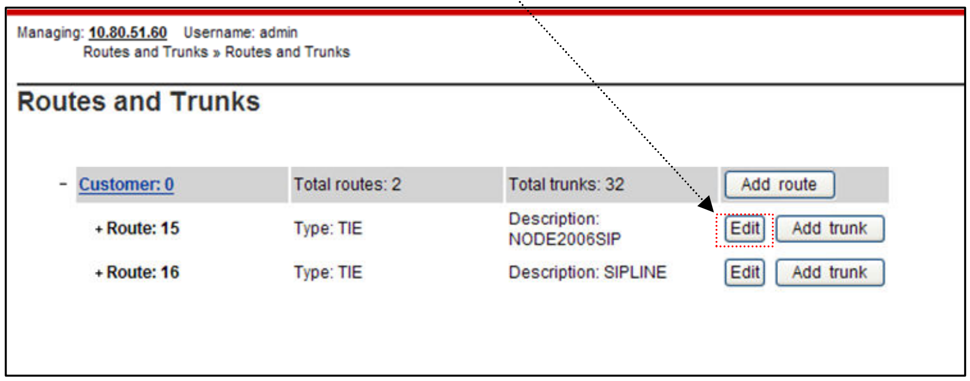
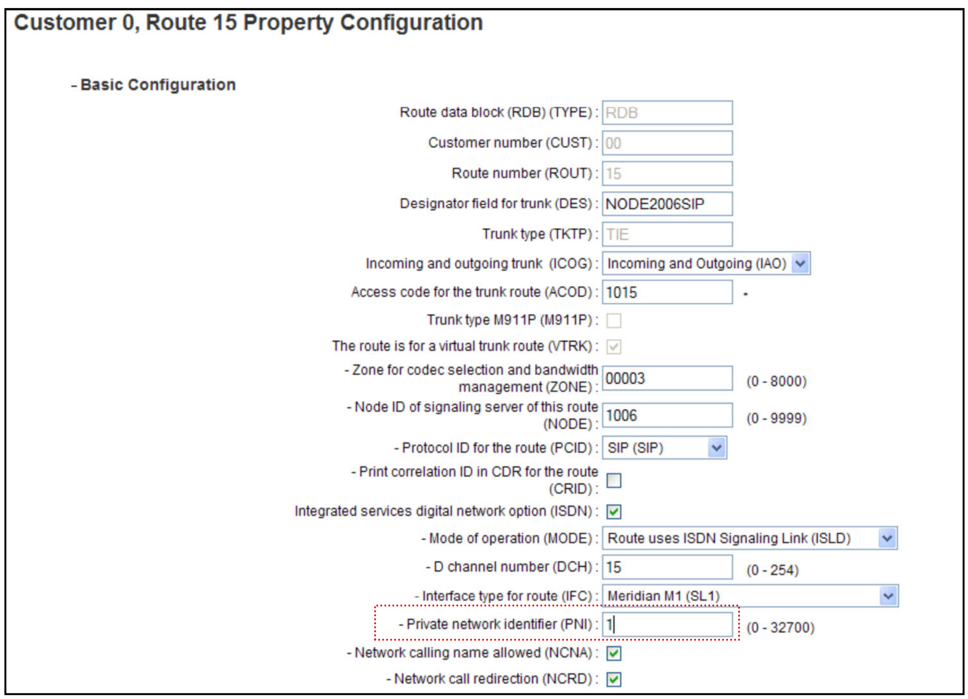
- Click on the **Edit** button on the right side that is associated with the **Route: 15**.

(This was the Route ID we defined for the virtual D-Channel – see pages 11 and 12)



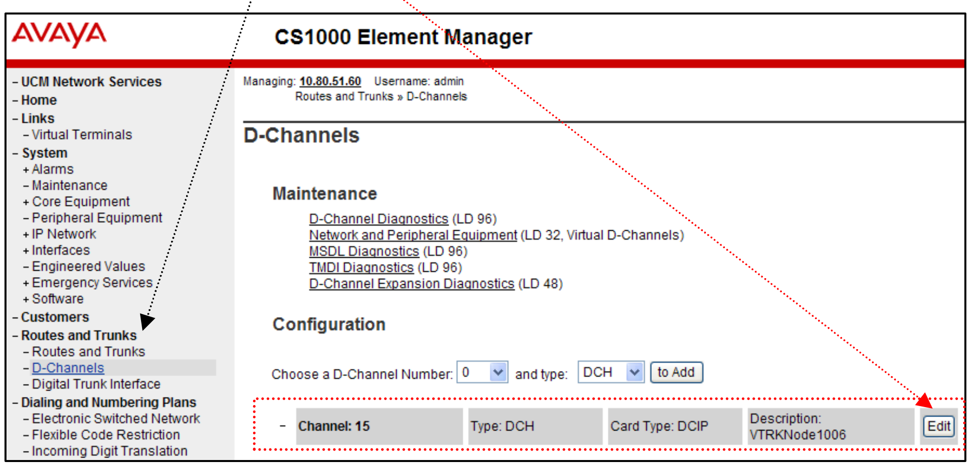
- The **Basic Configuration** section of Route 15 is now displayed.
  - In the **Private network identifier (PNI)** field, enter **1**

(This was the PNI you defined in the ISDN section – see page 19)

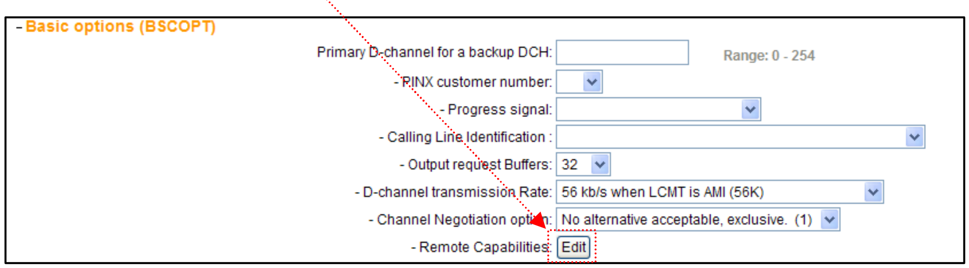


- Click **Commit** *(not shown)* to save changes.

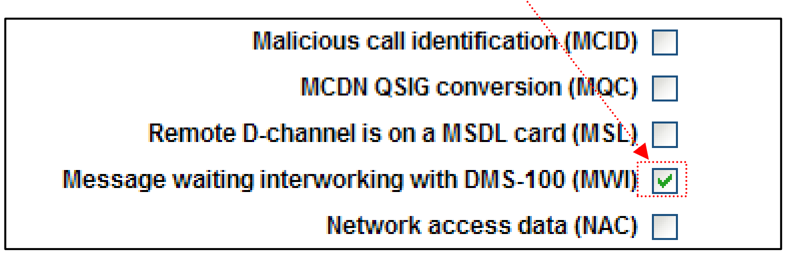
- **Verify MWI is enabled**
  - On the left side expand see **Routes and Trunks.**
  - Select **D-Channels**.
  - You will once again see the D-channels you administered
  - In our configuration we used Channel 15 as our D-Channel and the Card Type is **DCIP**, noting the Channel is a *virtual* (IP) D-channel.
  - Click the **Edit** button associated with this virtual D-Channel



- On the next screen expand **Basic Options (BSCOPT)**
- Click **Edit** next to **– Remote Capabilities**



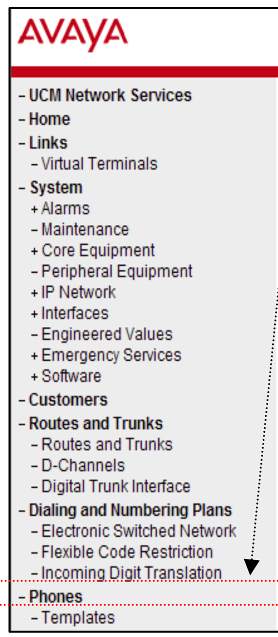
- On this next screen ensure **Message waiting interworking with DMS-100 (MWI)** is enabled.



5.2 SUBSCRIBER ADMINISTRATION

Subscriber administration includes:

- Configure Phones to cover to the Aura Messaging 'pilot' extension
- Every Aura Messaging subscriber's station/phone on the CS1000E will need to be configured with the 'pilot' number of 4445000 so that **busy** and **no-answer** calls will route to Aura Messaging. Although there are a number of tools that for telephone administration on the CS1000E (*i.e, Element Manager, Telephony Manager, and the command-line overlay terminal*) for this document we will continue to use Element Manager to administer the telephones.
- From the left-pane of Element Manager select **Phones**.
    - Use the **Search Criteria** to select the station to edit a station. The Value field is where you enter your station number.



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In **Features** table, scroll to the **Call Forward** features

Set the following using the drop down menus

- **CFHA – Call Forward/Hunt Override** = **Allowed**
- **CFXA – Call Forward External** = **Allowed**

Feature	Description	
CFHA	Call Forward/Hunt Override	Allowed
CFTA	Call Forward by Call Type	Denied
CFXA	Call Forward External	Allowed

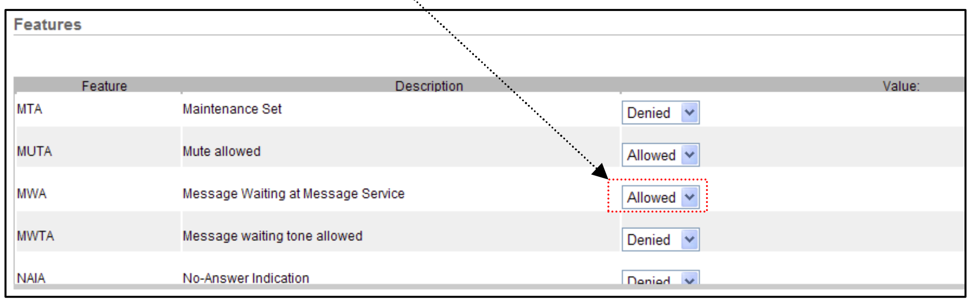
- Now scroll to the **FDN – Flexible Call Forward No Ans DN**
- In the adjacent field enter the Pilot Number, 4445000, for Avaya Aura® Messaging

Feature	Description	
FCAR	Force Charge Account	No
FDN	Flexible Call Forward No Ans DN	4445000

- Now scroll to the **HTA – Hunting** and set the following:
  - **HTA – Hunting = Allowed**
  - **HUNT – Hunt DN – All Calls, or Internal Calls for CFTA =** Enter the Pilot Number, 4445000, for Avaya Aura® Messaging.

Feature	Description	
HTA	Hunting	Allowed
HUNT	Hunt DN - All Calls, or Internal Calls for CFTA	4445000

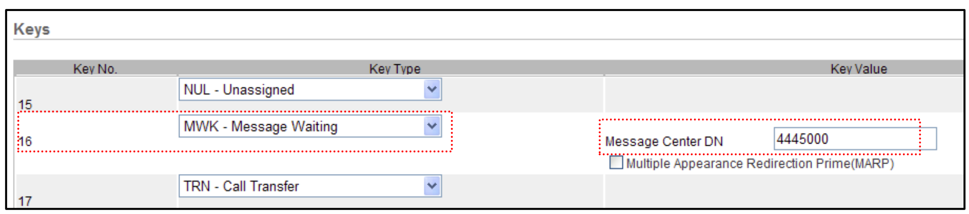
- Now scroll to the **MWA – Message Waiting at Message Service**
- Set it to **Allowed**



Features

Feature	Description	Value:
MTA	Maintenance Set	Denied
MUTA	Mute allowed	Allowed
MWA	Message Waiting at Message Service	Allowed
MWTA	Message waiting tone allowed	Denied
NAIA	No-Answer Indication	Denied

- Go to the **Keys** table
- Scroll down to Key No. **16**
- Use the adjacent drop-down menu in the Key Type column and select “**MWK – Message Waiting**
- In the **Message Center DN** field, enter the Pilot Number of 4445000 once again for Avaya Aura® Messaging.
- Click Save (not shown) to save your work.



Keys

Key No.	Key Type	Key Value
15	NUL - Unassigned	
16	MWK - Message Waiting	Message Center DN 4445000 Multiple Appearance Redirection Prime(MARP)
17	TRN - Call Transfer	

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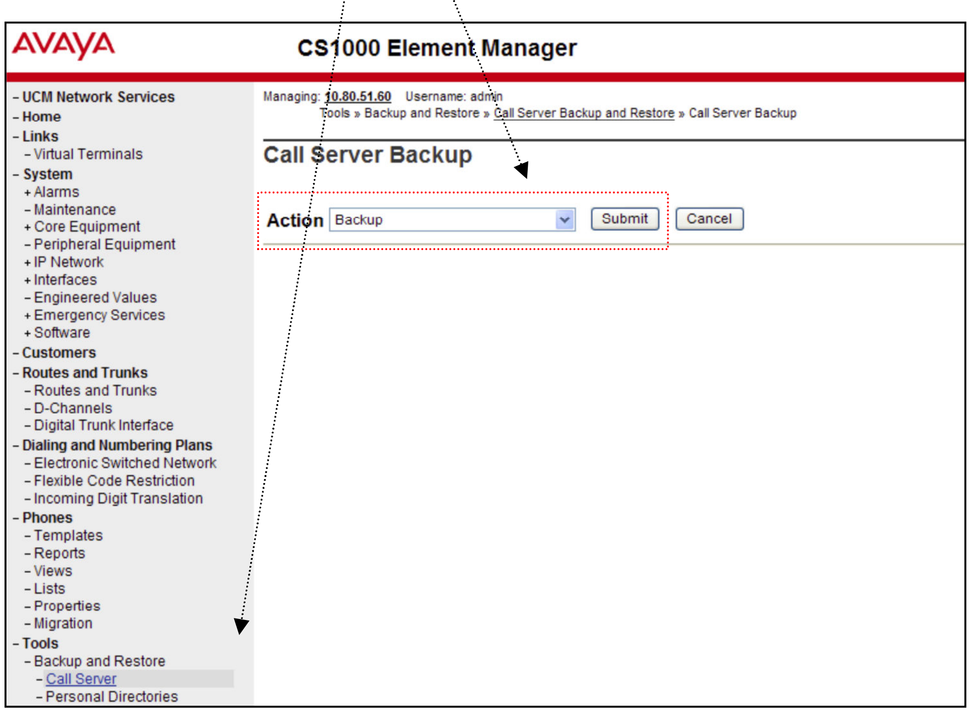
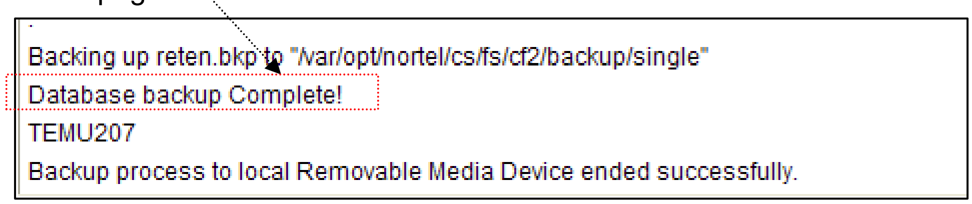
- Ensure you SAVE your configuration by doing the following
  - Expand **Tools**, then expand **Backup and Restore**
  - Select **Call Server**, then **Backup**
  - You will now see the **Call Server Backup** screen (shown below)
  - On the right side select the **Action** Backup
  - Then click **Submit** to save configuration



- The Backup process will take several minutes to complete.
- To see when the backup is complete, scroll to the bottom of the page.



- This completes your configuration of your CS1000

Important:

The domain names, IP addresses, etc. provided in this document are only examples.

Your domain names, IP addresses, etc. may be different.

5.4 CONFIGURING THE AVAYA AURA SESSION MANAGER

Avaya Aura Session Manager routes calls between the Avaya CS1000E and Avaya Aura Messaging.

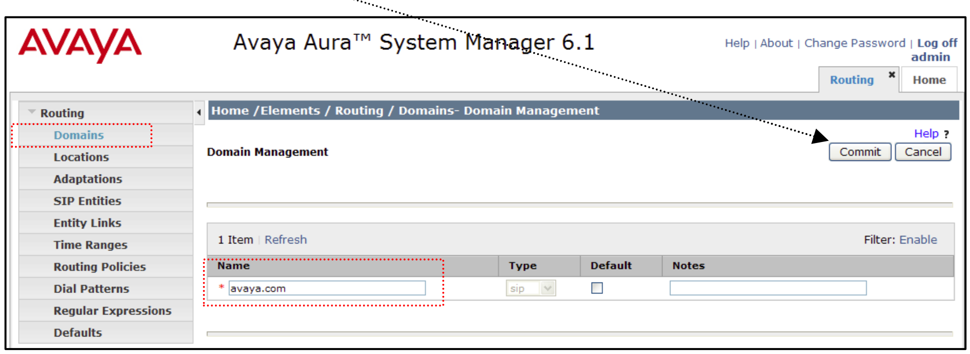
Assumptions:

- SIP entities for Avaya Communication Server 1000E and Session Manager are defined.
- Network connections are defined for:
  - Link between the Avaya System Manager and Avaya Session Manager
  - SIP trunk between Avaya CS1000E and Avaya Session Manager.

Most administration on Avaya Aura SM is done from the Network Routing Policy screens accessed from the Routing section on the left. For more complete programming information on Avaya Aura Session Manager please refer to the appropriate documentation.

5.4.1 Define a SIP Domain

- Expand **Elements,** then **Routing**
- Select **Domains** from the left navigation menu.
- Click **New** (not shown)**.**
- Enter the following values and use default values for remaining fields**.**
  - Name: **avaya.com** (our example Domain Name for the configuration.)
  - Type: **SIP**
  - Notes**:** <**a brief description** > (optional)
- Click **Commit** to save.



5.4.2 Define Location for Avaya Aura® Messaging

Locations identify logical and/or physical locations of SIP Entities. On the left side of your screen:

- Expand **Elements**
- Expand **Routing**
- Select **Locations**
- Click **New** (not shown)**.**

In the **General** section (shown below) enter the following:

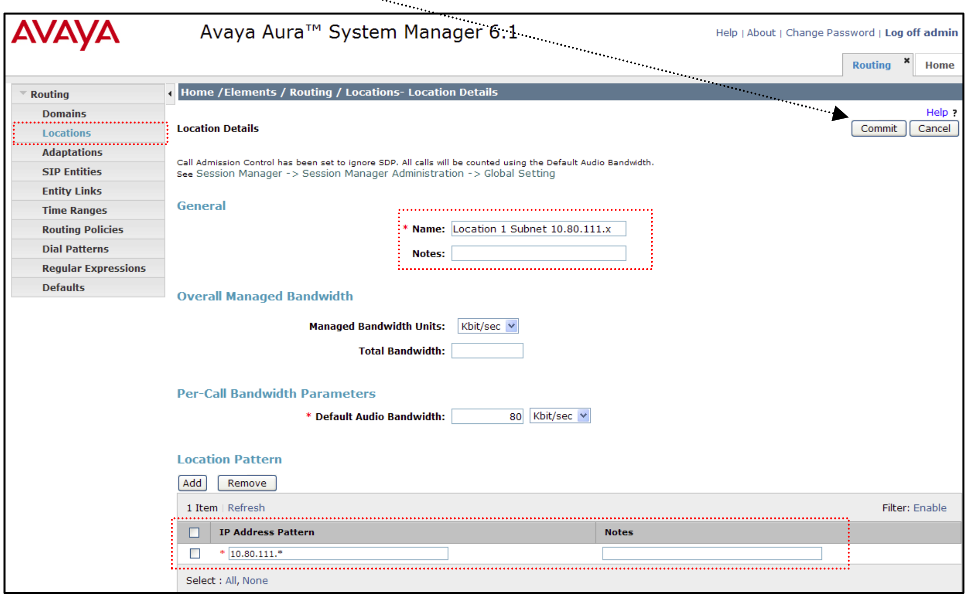
<u>Note</u>: Leave all fields/values not noted set to their default

- Name: <**a descriptive name for the location**>
- Notes: <**a brief description**> [Optional]

- In the **Location Pattern** section
  - Click **Add** and enter the following:
    - **IP Address Pattern:** < **the logical pattern used to identify the location**>

    <u>Note</u>: In our example we entered 10.80.111.*

  - Notes: <**a brief description**> [Optional]
- Click **Commit** to save.



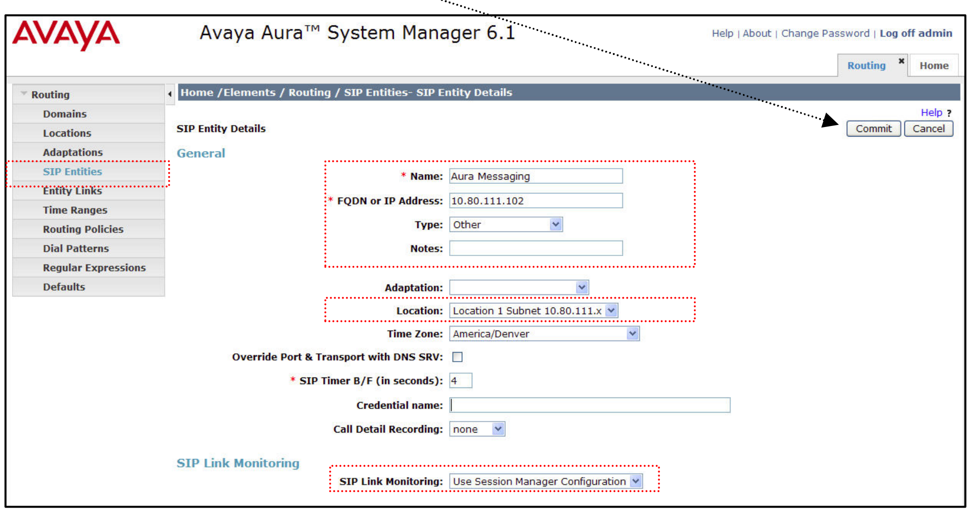
5.4.3 Define a SIP Entity

You need to define a SIP Entity for Avaya Aura Messaging. On the left side of your screen:

- Expand **Elements**
- Expand **Routing**
- Select **SIP Entities**
- Click **New** (not shown)
- In the **General** section (shown below) enter the following:

  Note: Leave all fields/values not noted set to their default

  - Name: <**a descriptive name for the entity** >
  - Notes: <**a brief description**> [Optional]
  - FQDN or IP Address: **<**Enter IP address of the Avaya Aura Messaging>
  - Type: "**Other**"
  - **Notes:** <**a brief description**> [Optional]
  - **Location: <**select the Location defined for Avaya Aura Messaging. This is the name you used for Location Details in the previous section. See 5.4.2>
- In the **SIP Link Monitoring** section (below) enter the following
  - SIP Link Monitoring: Select **Use Session Manager Configuration**
- Click **Commit** to save



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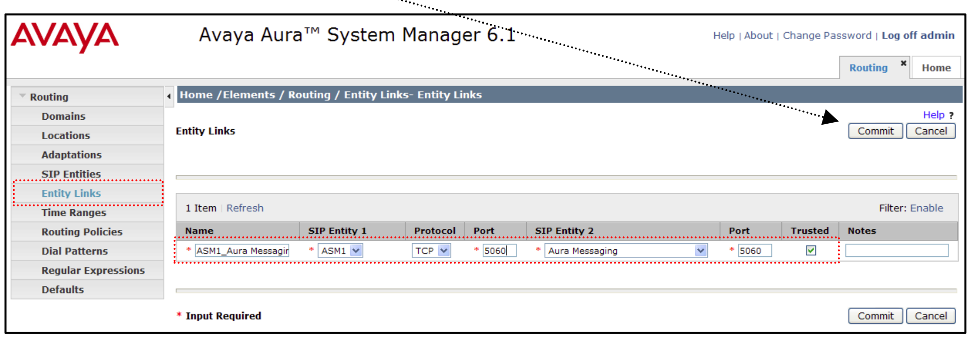
5.4.4 Define a SIP Entity Link

You now need to define the SIP Entity Link between the Avaya Aura Session Manager and Avaya Aura Messaging. On the left side of your screen:

- Expand **Elements**
- Expand **Routing**
- Select **Entity Links**
- Click **New** (not shown) and enter the following:
  - Name: <**a descriptive name for the link**>
  - SIP Entity 1: **ASM1** <this is the SIP Entity defined for Avaya Session Manager>
  - SIP Entity 2: **Aura Messaging** <this is the SIP Entity defined for Avaya Aura Messaging. This is the name you set in SIP Entity Details in the previous section 5.4.3>
  - Protocol: **TCP** <TCP was used in our sample configuration , but TLS is also an option that could be used. TLS provides security.>
  - Port: **5060** <TCP uses port 5060; TLS uses port 5061>
  - Trusted: **check** the box to set as a trusted entity
  - Notes: <**a brief description**> [Optional]
- Click **Commit** to save



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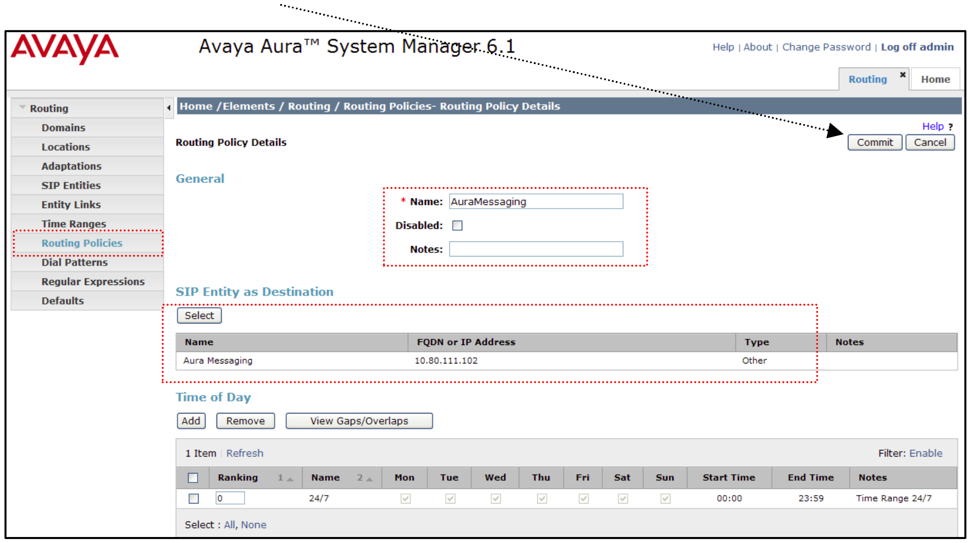
5.4.5 Define Routing Policy for the Avaya Aura Messaging

Routing policies specifies the rules (policies) used to route between the Avaya CS1000 and Avaya Aura Messaging. On the left side of your screen:

- Expand **Elements**
- Expand **Routing**
- Select **Routing Policies**
- Click **New** (not shown)
- In the **General** section (shown below) enter the following:

  Note: Leave all fields/values not noted set to their default

  - Name: <**a descriptive name/identifier for this Routing Policy**>
  - **Disabled: <**Leave unchecked>
  - Notes: <**a brief description**> [Optional]
- In the **SIP Entity as Destination** section
  - Click **Select**
  - The **SIP Entity List** will now display (not shown).
  - Select the **Name** of the SIP Entity used for Avaya Aura Messaging. <this is the SIP Entity defined for Avaya Aura Messaging. This is the name you set in SIP Entity Details in the previous section 5.4.2>
  - Click **Select**.
  - The **Routing Policy Details** is now displayed for Avaya Aura Messaging (see screen below)
- Click **Commit** to save



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5.4.6 Define Routing Policy for the Avaya CS1000E

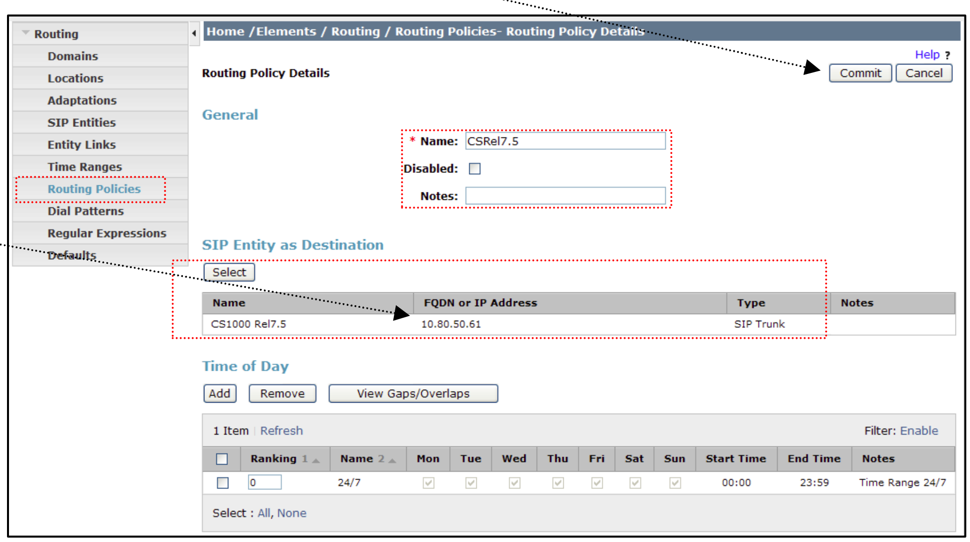
Now you repeat the steps as in the previous section 5.4.3. to define a Routing Policy for the CS1000E. On the left side of your screen:

- Expand **Elements**
- Expand **Routing**
- Select **Routing Policies**
- Click **New** (not shown)
- In the **General** section (shown below) enter the following:

  Note: Leave all fields/values not noted set to their default

  - Name: <**a descriptive name/identifier for this Routing Policy**>
  - **Disabled: <**Leave unchecked>
  - Notes: <**a brief description**> [Optional]
- In the **SIP Entity as Destination** section
  - Click **Select**
  - The **SIP Entity List** will now display (not shown).
  - Select the **Name** of the SIP Entity used for Avaya CS1000E <this is the SIP Entity that we assumed was already defined for the Avaya CS1000 that you are integrating with. If this Entity is not defined you will need to add it following the same procedure when you added the Avaya Aura Messaging as a SIP Entity. See section 5.4.2>
  - Select the **Name** of the SIP Entity used for Avaya Aura Messaging. <this is the SIP Entity defined for Avaya Aura Messaging. This is the name you set in SIP Entity Details in the previous section 5.4.2>
  - Click **Select**.
  - The **Routing Policy Details** is now displayed for Avaya Aura Messaging (see screen below)
- Click **Commit** to save

Note: The IP address 10.80.50.61 is the **Node TLAN IP Address** in **Node Details in Section 5.1** (see page 9 in this Config Note)



5.4.7 Define Dial Pattern for the stations

In our configuration we had defined two dial patterns for routing calls between the Avaya CS1000E and Avaya Aura Messaging.

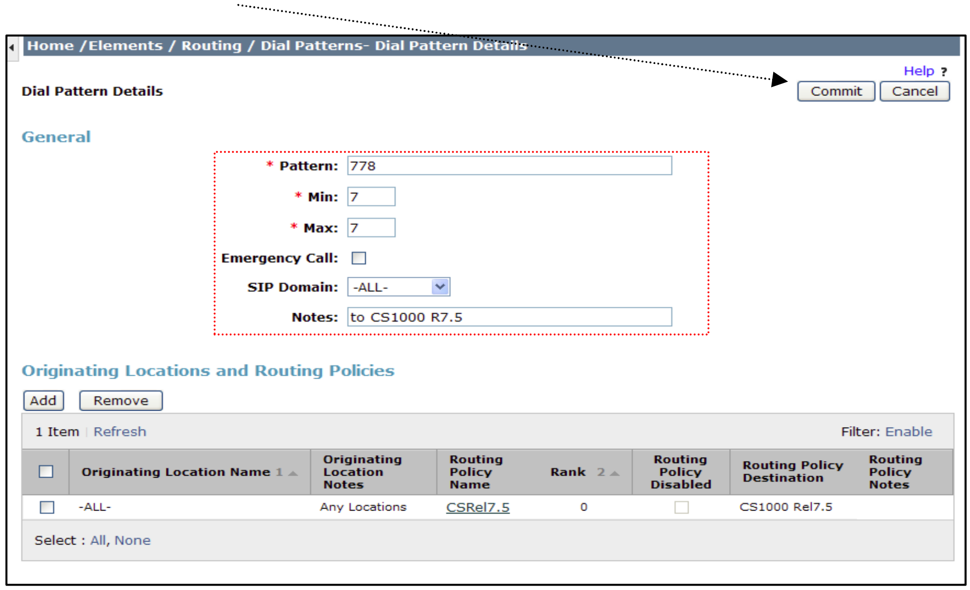
- 778 was defined for station numbering plans
- 4445 was defined for the Pilot Number

To define a Dial Pattern from the left side of your screen:

- Expand **Elements**
- Expand **Routing**
- Select **Dial Patterns** (not shown)
- Click **New** (not shown).
- In the **General** section, enter the following:

  Note: Leave all fields/values not noted set to their default

  - **Pattern: 778** <this is the dial pattern for the stations>
  - **Min: 7 <** this is the minimum number digits required to dial>
  - **Max:** <Enter the maximum number digits that may be dialed>
  - **SIP Domain: ALL** <select the specific SIP Domain from drop-down menu or ALL if Session Manager should accept incoming calls from all SIP Domains>
  - Notes: <**a brief description**> [Optional]
- In the **Originating Locations and Routing Policies** section
  - Click **Add**
  - The **Originating Locations and Routing Policy List** is now displayed (not shown).
  - In the **Originating Locations** table select **ALL**
  - Notes: <**a brief description of the Dial Pattern**> [Optional]
  - In the **Routing Policies** table, select the Routing Policy defined for Avaya Communication Server 1000E in **Section 5.4.6**
  - Click **Select** to save your changes and return to **Dial Pattern Details** page.
- Click **Commit** to save



5.4.8 Define Dial Pattern for the Pilot Number

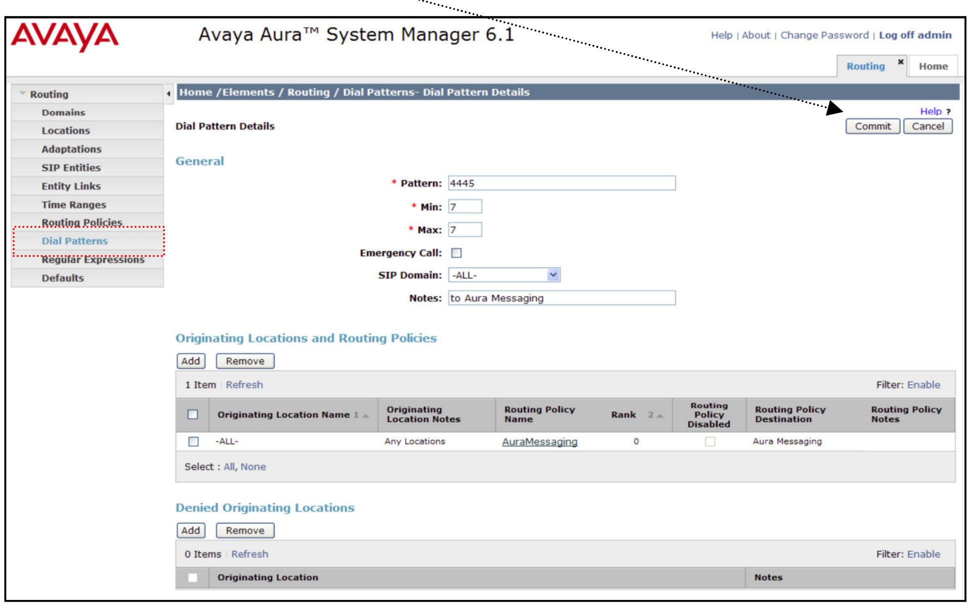
Here we repeat the steps as in section 5.4.7 to define the dial pattern to route calls to the Pilot Number for Avaya Aura Messaging.

To define this Dial Pattern from the left side of your screen:

- Expand **Elements**
- Expand **Routing**
- Select **Dial Patterns** (not shown)
- Click **New** (not shown).
- In the **General** section, enter the following:

  Note: Leave all fields/values not noted set to their default

  - **Pattern: 4445** <this is the dial pattern for the Pilot Number>
  - **Min: 7 <** this is the minimum number digits required to dial>
  - **Max:** <Enter the maximum number digits that may be dialed>
  - **SIP Domain: ALL** <select the specific SIP Domain from drop-down menu or ALL if Session Manager should accept incoming calls from all SIP Domains>
  - Notes: <**a brief description**> [Optional]
- In the **Originating Locations and Routing Policies** section
  - Click **Add**
  - The **Originating Locations and Routing Policy List** is now displayed (not shown).
  - In the **Originating Locations** table select **ALL**
  - Notes: <**a brief description of the Dial Pattern**> [Optional]
  - In the **Routing Policies** table, select the Routing Policy defined for Avaya Communication Server 1000E in **Section 5.4.6**
  - Click **Select** to save your changes and return to **Dial Pattern Details** page.
- Click **Commit** to save



6.0 CONFIGURING THE AURA MESSAGING SERVER

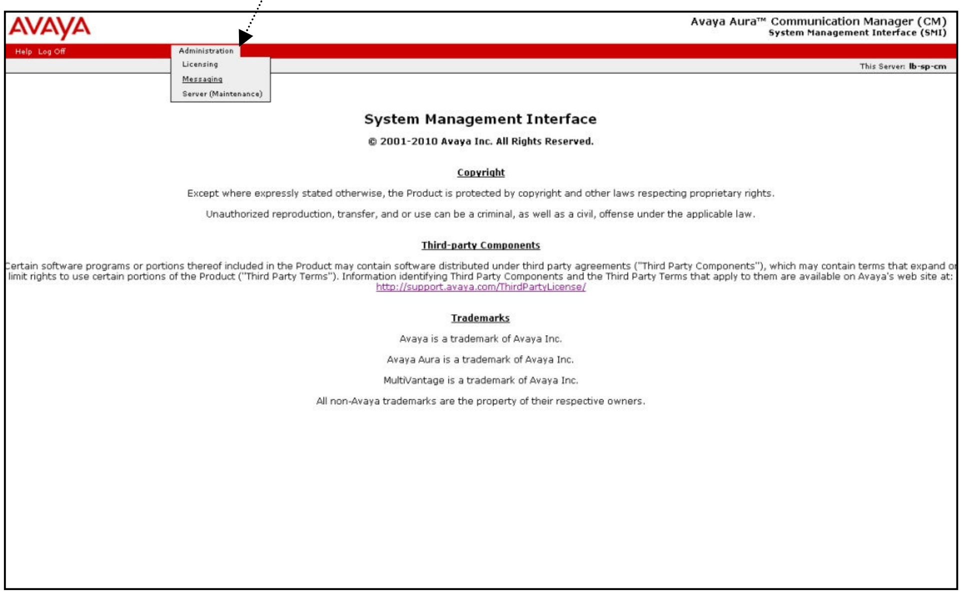
Configuring the Aura Messaging Server

The procedures to complete the integration of Avaya Aura Messaging will require the following Administration:

- Adding Site (Avaya Aura Messaging Server)
- Configure the Telephony Integration

6.0.1 Administer Messaging

- Select **Administration**
- Select **Messaging**



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6.0.2 Add Site (Avaya Messaging)

On the left side of the screen under Messaging System (Storage):

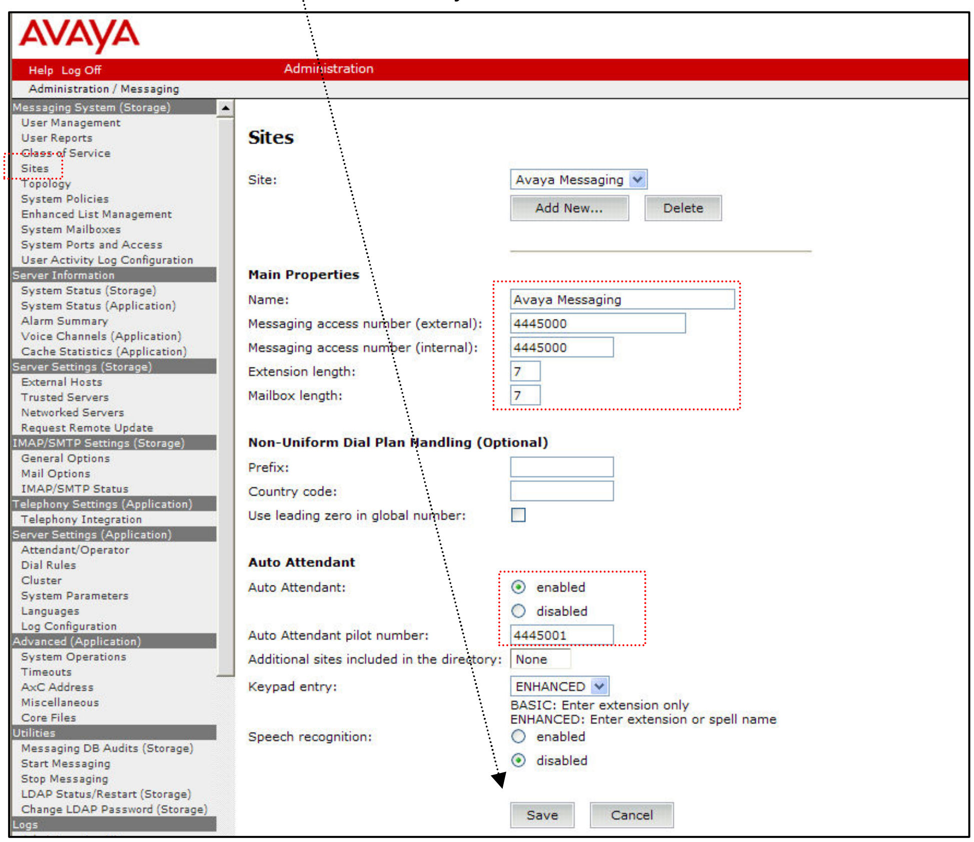
- Select **Sites**

The Sites screen will display (shown below). On the right

- In the **Sites** section under **Main Properties**, enter the following:
  - Name: **Avaya Messaging** <a descriptive name/identifier for this site>
  - Messaging access number (external): **4445000** <the Pilot number for the site>
  - Messaging access number (internal): **4445000** <the Pilot number for the site>
  - Extension Length: **7** <the number of digits in the station numbers>
  - Mailbox Length: **7** <the number of digits in the mailbox numbers>
- In the **Sites** section under **Auto Attendant**:
  - Auto Attendant: **Enable** <click on the radio button to enable>
  - Auto Attendant Pilot Number: **444-5001** <the Auto Attendant number for the site>
  - Click **Save** to save your site definition

Note:
In our configuration we used -
444-5000 as our Pilot #
- and -
444-5001 as our Auto Attendant #

The Pilot and Auto Attendant Numbers in your configuration may be different.



6.0.3 Administering the Telephony Integration

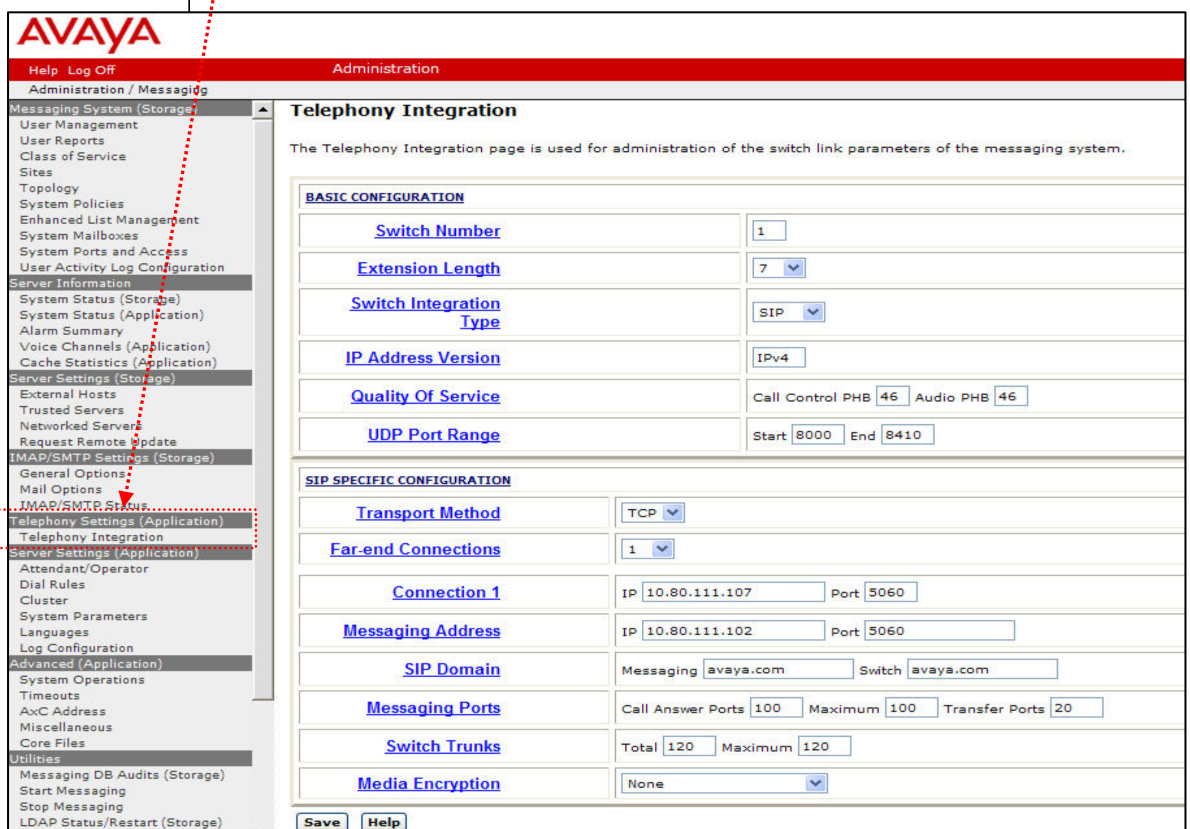
- In the **Administration / Messaging** menu on the left side, under **Telephony Settings (Application)**
  - Select **Telephony Integration**

The **Telephony Integration** Screen will display (shown below).

There are two sections to this screen that will be used to complete the Telephony Integration of Avaya Aura Messaging. They are:

- **Basic Configuration**
- **SIP Specific Integration**

These sections will be detailed in the next pages/sections

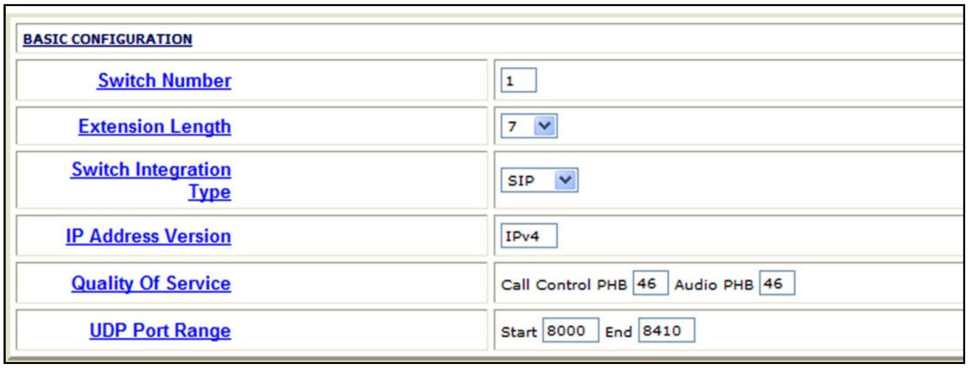


Note:

Please note, the settings shown in Basic and SIP Specific Configuration are for example only. Your settings may be different.

6.0.3.1 Basic Configuration

The **BASIC CONFIGURATION** section of the Telephony Integration Screen is shown below. Please refer to the configuration details shown below the screen to administer your site.



BASIC CONFIGURATION (Parameters/Settings):

- **Switch Number = *1*** <always set to 1 unless Avaya Support directs otherwise>
- **Extension Length = 7** <*extension length up to 10 digits*>. The extension length of 7 in the screen matches the dial plan of the media server.
- **Switch Integration Type** = *SIP*
- **IP Address Version** = Ipv4
- **Quality of Service** = A Value of *0* to *63* may be used.

The Call Control PHB and Audio PHB set the QOS levels for call control and audio messages on networks that support this feature.

These values must match the corresponding numbers on page 1 of the IP Network Region screen (see IP Network Region screen in Section 5.1 in this CN) under the DIFFSERV/TOS PARAMETERS. If numbers do not match then call failures may result.

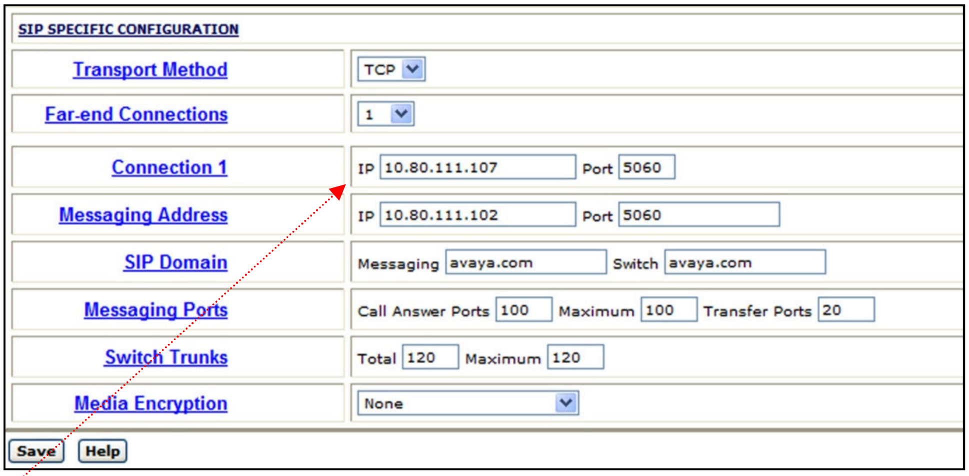
- **UDP Port Range** – **Start** = 8000 **/ End** = 8410

Note:

Please note, the settings shown in Basic and SIP Specific Configuration are for example only. Your settings may be different.

6.0.3.2 SIP Specific Configuration

The **SIP SPECIFIC CONFIGURATION** section of the Telephony Integration Screen is shown below.



** IMPORTANT **
Encryption (TLS & SRTP) are currently not supported. Please disable "msec" (media security) or leave to "Best Effort" in which case AAM & CS1K will negotiate down to no security.

SIP SPECIFIC CONFIGURATION (Parameters/Settings):

- **Transport Method** = *TCP* or *TLS**. <This is the transport method used for SIP signaling and must match the transport method administered on the switch>
- **Far-end Connections** = *1.* <This is the number of far-end connections to administer>
- **Connection 1** = *10.80.111.102* <This is the IP Address of the Aura Session Manager>
  - Port number = 5060 (*5060 is* for *TCP ; 5061* for *TLS)*
- **Messaging Address** = *10.80.111.102* <This is the IP address of Aura Messaging as defined in section 5.4.3>
  - Port number = 5060 (*5060 is* for *TCP ; 5061* for *TLS)*
- **SIP Domain = <***domain name*> <This is the Domain Name we used for our configuration, see Section 5.4>
- **Messaging Ports –**
  - **Call Answering Ports** = 2 or more. <The number of call answering ports configured on the system. This could be less than or equal to the maximum number of ports available
  - **Maximum** = xxx <The **maximum** number of ports that may be configured as Call Answering ports>.

** IMPORTANT **
Encryption (TLS & SRTP) are currently not supported. Please disable “msec” (media security) or leave to “Best Effort” in which case AAM & CS1K will negotiate down to no security.

  - **Transfer Ports** = **xx** <This field is read only and shows the ports available for transfer ports. This is calculated as the difference between the number of trunks and call answer ports>
- **Switch Trunks = xxx** <Must match the number of trunks configured for the messaging on the switch. If multiple signal groups are administered, this number is the sum of all trunks in all groups>
- **Media Encryption** = ** *None*
  srtp-aescm128-hac80
  srtp-aescm128-hmac32

Chose the encryption (*one of the 3 choices noted above*) that matches what is administered in the IP Codec Set used for this integration.

Click **Save** to save all changes.

Once this is done, the Switch Link Administration screen will be displayed notifying the user if a restart is required (sometimes a restart is not needed) for changes to take effect.

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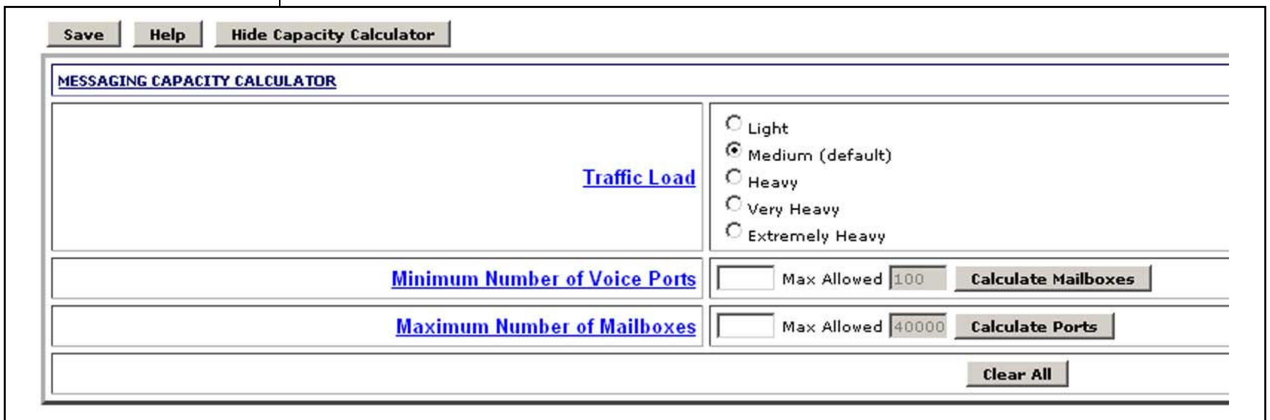
p.4- MESSAGING CAPACITY CALCULATOR

The button labeled “Show Capacity Calculator” just below the SIP SPECIFIC CONFIGURATION section is a tool that can be used to determine the number of call answer ports needed. If you click on *Show Capacity Calculator* the following screen appears.



HOW TO USE THE CALCULATOR

Traffic Load – Chose a traffic load profile that suits your needs. Table A (below) is a guideline to help determine traffic load.

Traffic Load	Voice Port Usage in Minutes (per subscriber per day)	Number of Voice Messages (per subscriber per day)
Light	2	1.5
Medium	4	3
Heavy	6	4.5
Very Heavy	8	6
Extremely Heavy	10	7.5

Table A. Traffic Load Guide

- **Minimum Number of Voice Ports - <**Enter the number of call answer ports (must be at least 2) >
- Click on the ***Calculate Mailboxes*** button to display the number of mailboxes (as recommended by Avaya)

- **Maximum Number of Mailboxes**[1] – Enter the number of mailboxes (must be at least 2).
- Click on the ***Calculate Ports*** button to display the number of call answer ports <as recommended by Avaya>

[1] *Maximum number of mailboxes is determined by your license.*

** IMPORTANT **
Encryption (TLS & SRTP) are currently not supported. Please disable "msec" (media security) or leave to "Best Effort" in which case AAM & CS1K will negotiate down to no security.

8.0 CONSIDERATIONS / ALTERNATIVES

8.1 **SIP integrations may not be reliable for TTY/TDD** if the IP network is unable to support uncompressed audio with no packet loss. For this reason **Avaya does not support TTY/TDD with this SIP integration**.

8.2 In reference to supported "transport CODECs", AAM supports only G.711. Ensure the far end SIP end point (SIP gateway or SIP PBX) is set accordingly. Failure may result in undesirable or what's perceived to be a non-working or dysfunctional AAM. G.711 is the front-ended transport CODEC, AAM's back-end storage allows for both GSM and G.711 CODECs to the message store. This latter switch setting is found within the SMI of "System Parameters".

8.3 **If your integration is set to use TLS as the transport method/link type and calls are not completing but they do complete using TCP,** then the cause may be a license issue.

8.4 Avaya Aura Messaging currently does not support E.164 formatted numbering or any mailbox or extension number exceeding 10-digits. You may need to add an Adaptation Rule to Session Manager to add/subtract digits or the Routing Pattern entry to handle this.

8.5 If you are using Outlook and attempt to Play a message on a phone that requires an outside trunk and the call get rejected/fails, check to see if service provide is blocking calls with names.

8.6 In a **network consisting of an Avaya CM and CS1000** with a Session Manager, if a call originates from a station on CM to a station on the CS1000, and subsequently gets transferred to another station on the same CS1000 (for example in a zero out scenario) the caller may experience **no talk path**. The workaround for this issue is to disable a feature in the CM SIP trunk-group called Network Call Redirection (NCR).

8.7 It's important to note special CS1K phone terminal PBX programming is warranted when having multiple DN extensions to a singular physical phone (i.e. main is 7000 (secretary) with execs on 7001, 7002 and 7003).

Real CLID values must be put into the configuration of the additional keys on the target set configuration. If you configure a "D" instead of a real CLID entry, it uses the key with CLID entry before the "D" as the digits that go to Aura Messaging. For example if key 0 is ext 7000 with a CLID 0 and key 1 is ext 7001 and has CLID D, 7001 will go to mailbox 7000. If you change 7001 to have a CLID 0 or any other number it will go into its own mailbox.

For more detailed information, see CS1K documentation and search for keyword “KEY”. It’s switch syntax is “KEY xx aaa yyyy (cccc or D) zz..z”.

CHANGE HISTORY		
Version	**Issue Date**	**Reason for Change**
A	1/27/2012	Moving document to GA status. No content changes.
B	4/25/2012	Added a special note to section 8.0 to address multiple DNs (extensions) to one physical phone. Configuration clarification concern.
C	5/9/2012	Clarification under Section 8 regarding CODECs.
D	6/25/2012	Removed DRAFT note.
E	8/26/2014	Removed need for CS1K related Premium Feature set.

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ADDENDUM FOR AUDIOCODES GATEWAY INTEGRATIONS

This section contains information regarding Issues and Solutions found with AudioCodes Gateways integrations.

Note for AAM: Only AudioCodes firmware version 5.80A.xxx.xxx is supported.

1. Issue: FIND ME: On a Find Me call when the called party answers they hear four DTMF digits (A, B, C, D) are played followed by about 1 second of silence, followed by the normal prompt with the first little bit missing).

SOLUTION: In the AudioCodes .ini file Add the *RxDTMFHangOverTime* parameter with a value of 100 instead of the default value of 1000ms.

2. Issue: *DTMF - User presses the # key in a recording which is translated to a slight "bleep" when the recording is listened to.*

SOLUTION: You can reduce the length of the DTMF chirp using a procedure for changing the recognition of DTMF in the AudioCodes. Please contact Integrations Support for this information.

3. Issue: *FAC -* Transfer to Voice Mail is a feature that is currently NOT SUPPORTED when using AudioCodes Gateways. A solution is currently under investigation.

4. Issue: *Transfer/FINDME Fails -* Calls originating through one Mediant Gateway to AAM, that have a new independent call established from the AAM through Mediant B will ring the end user but when call is answered user hears a tone and call is disconnected and a SIP 481 error is generated in the logs. Call is split and cannot be bridged as GWs do not know each has a leg of the same call.

SOLUTION: Use one Gateway. Multiple gateways are currently not supported

5. Issue: *Beep tone -* A beep tone is heard when on a transfer just before the Personal Greeting is played. On a RNA no tone is heard.

SOLUTION: This occurs because AAM sends an sdp with (audio) “a=inactive.” This then causes the Mediant gateway to play a HELP_TONE because it assumes that MoH (Music on Hold) will have to be played locally since there is no audio stream expected (a=inactive). The only way around this is to remove the tone from the CPT file in the Gateway. A CPT with this tone removed is available from Integrations Support.

6. Issue: *E1 calls fail on upper half of span -* If calls on E1 channels above 16 (the D-Channel for an E-1) have no talk path (dead air) it may be a setting in the AudioCodes Gateway causing it.

SOLUTION: In the AudioCodes ini file, check the ISDNGeneralCCBehavior parameter to see if it is set to 32. If so change it to 0, which is the default value. Then reload/burn the INI and calls should complete properly.