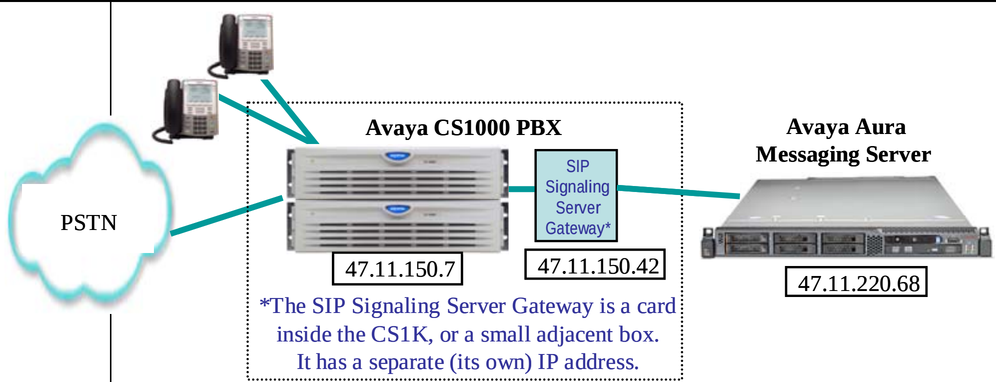
Avaya™

Aura Messaging

Configuration Note 88103 – Rev F (6/12)

Avaya CS1000

SIP Integration Directly to Avaya CS1000



Please refer to sections 2 and 3 in this Configuration Note for hardware and software requirements.

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 Config 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 trunks. This integration passes call information and MWI using SIP packets.

Avaya Aura Messaging Server Requirements

1 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.

PBX hardware requirements

PBX software requirements

2.0 AVAYA AURA MESSAGING SERVER REQUIREMENTS

- **Minimum Software releases required [1]:**
  - **Avaya Aura Messaging 6.0.1**
  - **RFUs (patches) to be determined**

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 Communication Server 1000 (CS1000) Rls 6.0
- Avaya Signaling Server for Direct Connect to Avaya Aura Messaging 6.00.18

3.1 PBX SOFTWARE REQUIREMENTS

Software [1 (see page 2)]:

- **Avaya CS1000 6.00R** (updated to the current DEPLIST).

  Note: "Direct Connect" was only tested under CS1K 6.0. Additional releases can be tested based on market demand/GRIP submissions.
- **Avaya Signaling Server for Direct Connect to Avaya Aura Messaging**
  - **Service Packs** – Latest
  - Nortel-cs1000-vtrk-6.00.18.65-90.i386.000.ntl or higher
- **VTRUNK SU** installed with the following activator patches are required
  - **MPLR29593_1** (activates support for UPDATE of p-assert after call answer)
  - **MPLR25946** (PI: SIP Line on CS1000 5.5: Remove MCDN from outgoing INVITE - "No Charge and No PLM approval required").

3.2 CONNECTIVITY

- Ethernet LAN connectivity – TCP/IP

3.3 CUSTOMER-PROVIDED EQUIPMENT

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

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 / Notify Me	[✓]
Direct Call	[✓]
External Call ID (ANI)	[✓]
Fax	[✓]
Find Me / Reach 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.

***Outcalling restrictions are determined by CS1000 administration and user COS**

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 the CS1000 has been completed.

For information on basic configuration please refer to *Communication Server 1000E Installation and Commissioning.* Release 6.0, rev 3.02. 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
- Add a route for the Aura Messaging 'pilot' extension

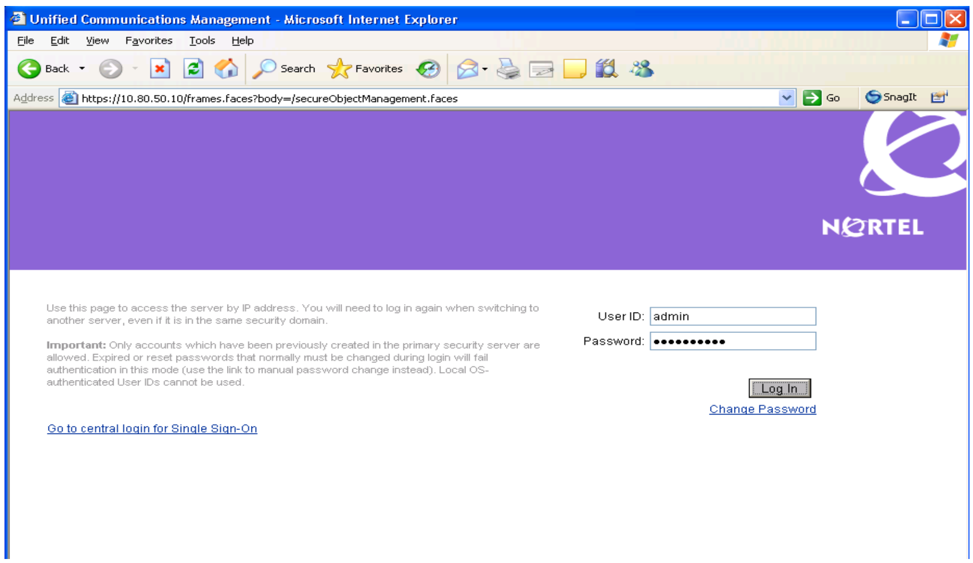
- continued on next page –

5.1 CONFIGURING THE AVAYA CS1000E USING THE IE BROWSER

- Open Internet Explorer and enter the IP Address of the CS1000E call server. In the example image below the URL to login is **https://10.80.50.10/**

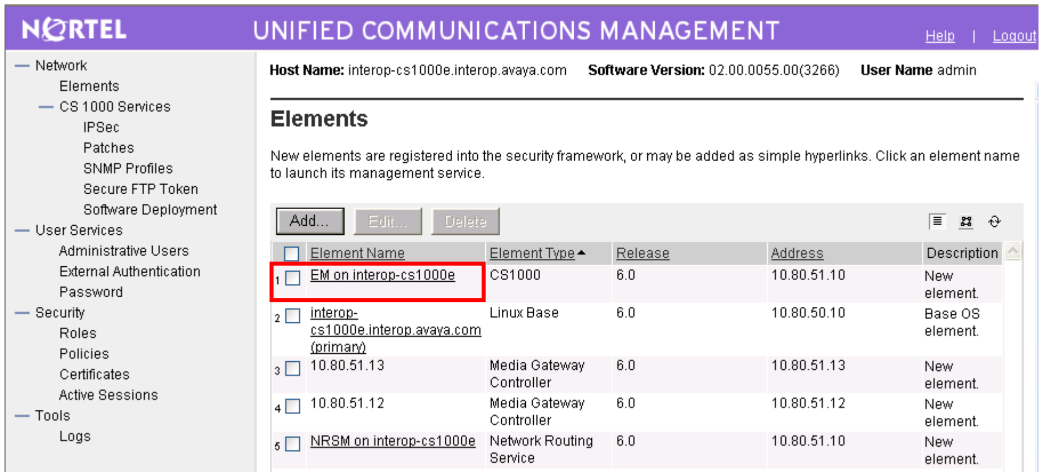
Note: IE is the only browser supported for CS1000E UCM

- This should bring you to the CS1000E Communications Management page.
- Log in using the appropriate Username and Password.



- continued on next page –

- Once logged in the first screen you will see is the Elements screen. Select the element of type **CS1000**.



- **ADD A DISTANT STEERING CODE (DSC)**

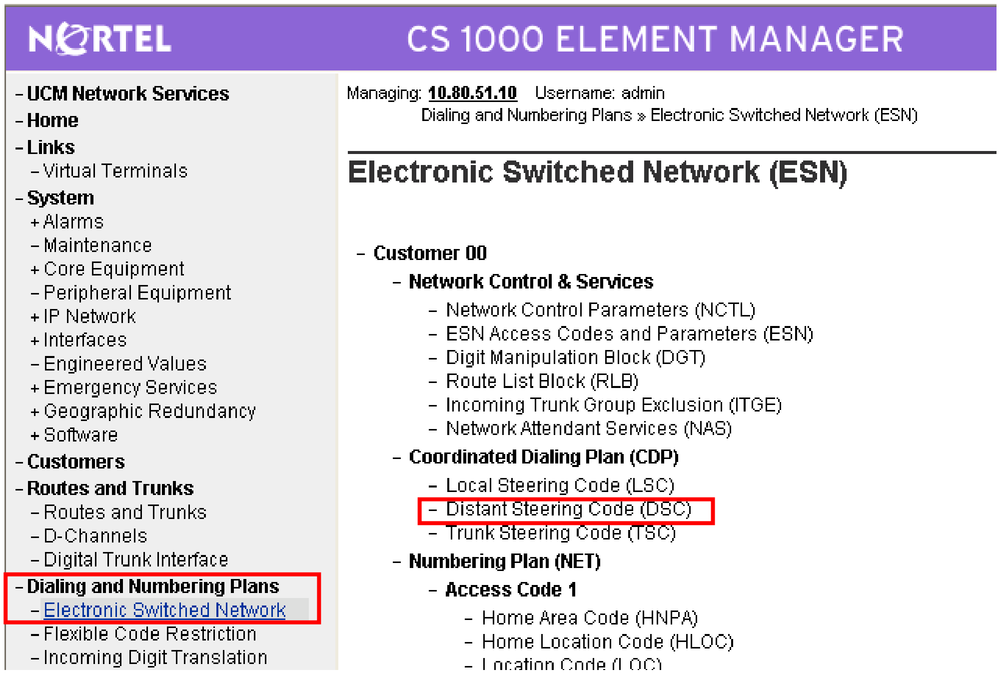
The CS1000E will router callers and subscribers to Aura Messaging using a Distant Steering Code, or DSC. In our example configuration the CS1000E only needs to route calls to CS1000 PBX, which will route the calls to Aura Messaging.

In this configuration, extension **2300** is our pilot number. This is the number used by subscribers to call to retrieve messages, and also the number that the CS1000E will use to cover to voice mail.

To do this we need to add a **Distant Steering Code (DSC)** for any number that starts with **230** and is 4-digits in length. This is shown on the next page. What this does is route any number you dial beginning with 230, for instance 2301, 2302, 2303, etc., to Avaya Aura Messaging.

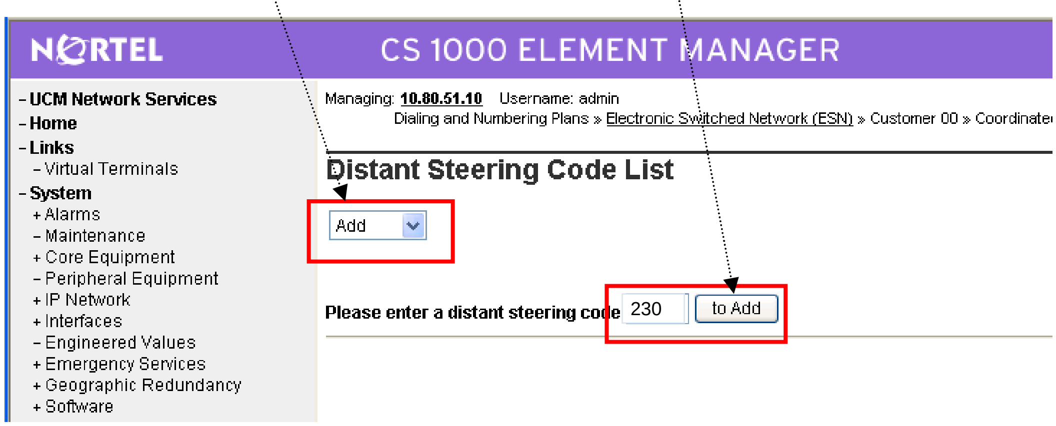
- continued on next page –

- To add a **DSC**, from the left-pane select **Electronic Switched Network**. Then, from the newly displayed right-panel select Distant Steering Code as indicated below.



- The screen below should now appear. Using the drop-down menu select **Add,** and then enter **230** in the field next to *Please enter a distant steering code*. Then click on "**to Add**"

<u>**NOTE**</u>: It's not necessary to differentiate all numbers that begin with **230.** It's only necessary to get those calls that have a number beginning with **230** to the Avaya CS1000.

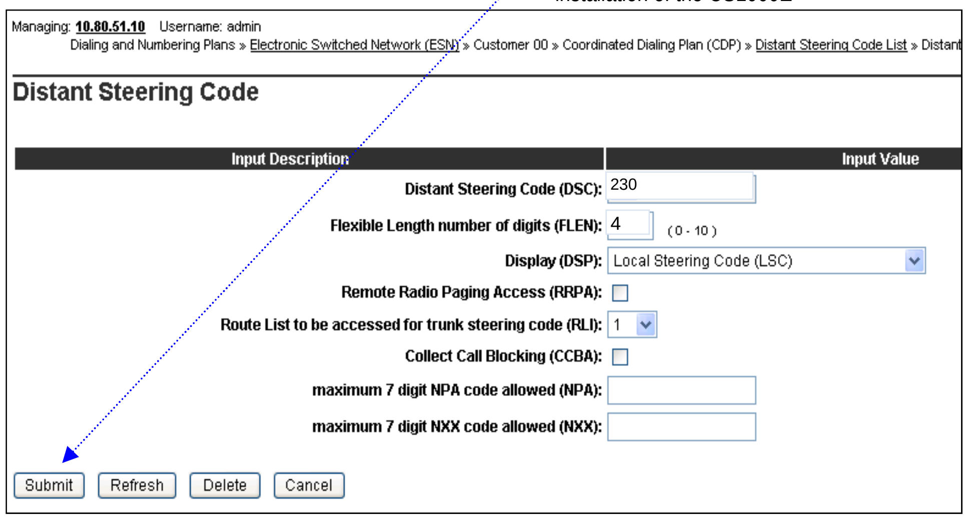


- The **Distant Steering Code** screen should now appear with **230** in the field adjacent **Distant Steering Code (DSC)**.
- Enter the following values and then click on **Submit**:

Flexible Length Number of digits (FLEN): 4 <*Maximum length of number starting with 2300*>

Display (DSP): Local Steering Code (LSC)

Route List accessed for trunk steering code (RLI): 1 <this is the Route List built between the CS1000E Call Server and Signaling Server. In our example, RLI 1 was configured during the installation of the CS1000E>



- continued on next page –

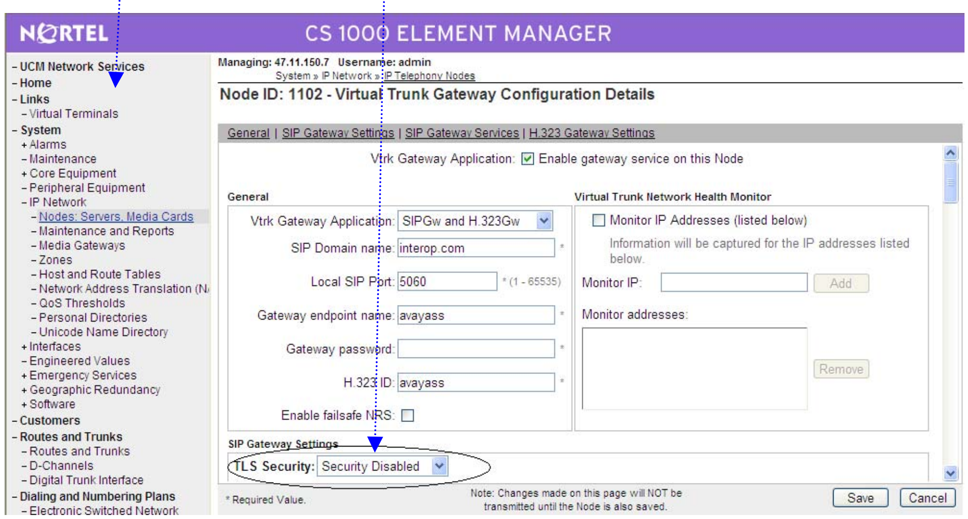
5.2 CONFIGURING THE CS1000 RLS 6.0 TRUNK GATEWAY

<u>NOTE</u>: A SIP SIGNALING SERVER GATEWAY IS REQUIRED TO CONNECT THE CS1000 DIRECTLY TO AVAYA AURA MESSAGING.

- From the UCM main menu, click on *<u>Nodes: Servers, Media Cards</u>* and select the SIP Gateway that connects to AVAYA AURA MESSAGING.
- Enter the following values. You will need to scroll down the screen to enter all values. Once complete click on **Save**:

SIP Domain: *your.domain.name* <In our example screen we show "*interop.com*" Please note that your domain may be different. Please consult with your company's network administrator>

TLS Security: Disabled

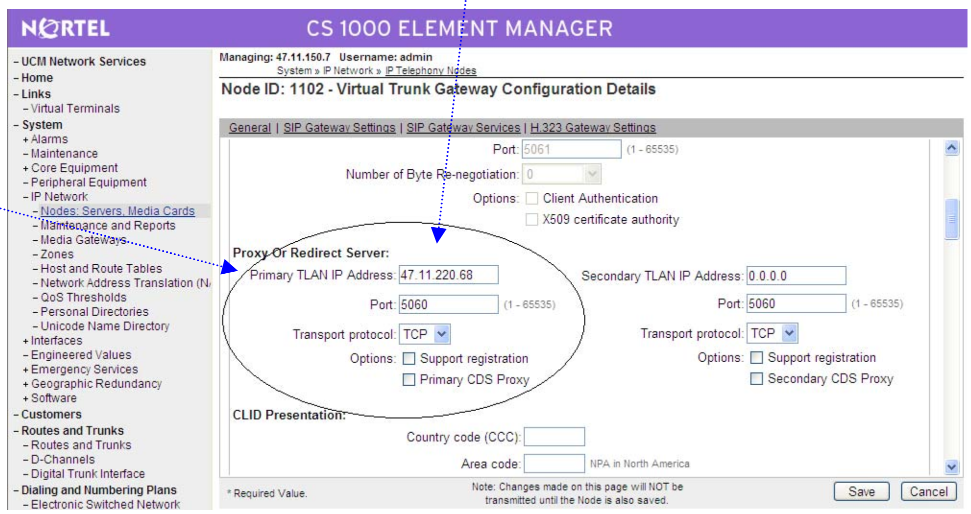


- continued on next page –

Primary TLAN IP Address: 47.11.220.68 <This is the IP address of your AVAYA AURA MESSAGING>

Port: 5060

Transport Protocol: TCP



Note: The IP address shown in the screen here in the Primary TLAN IP Address field is an IP address we used for our sample configuration. Your IP address will be different.

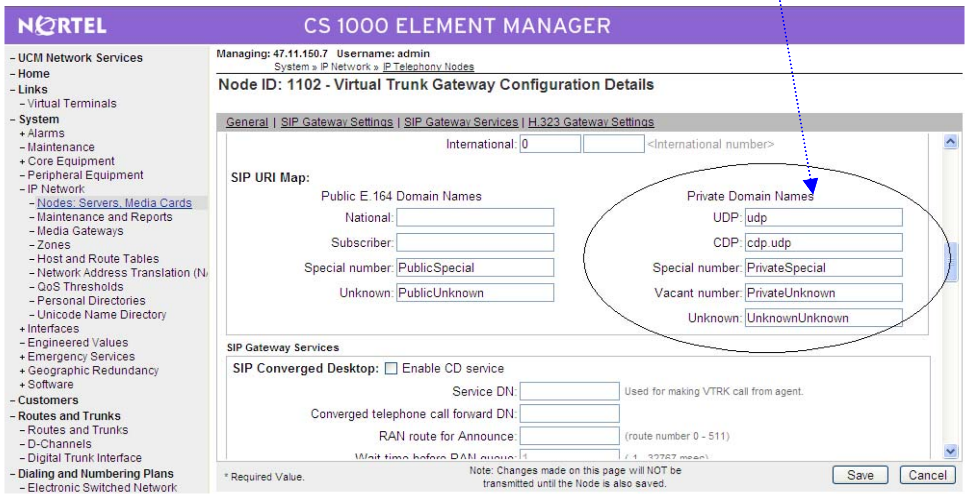
- continued on next page –

In the Private Domain names section (*circled below*)

UDP: Leave Blank. <The system will enter udp by default>

CDP: Leave Blank. <The system will enter cdp by default>

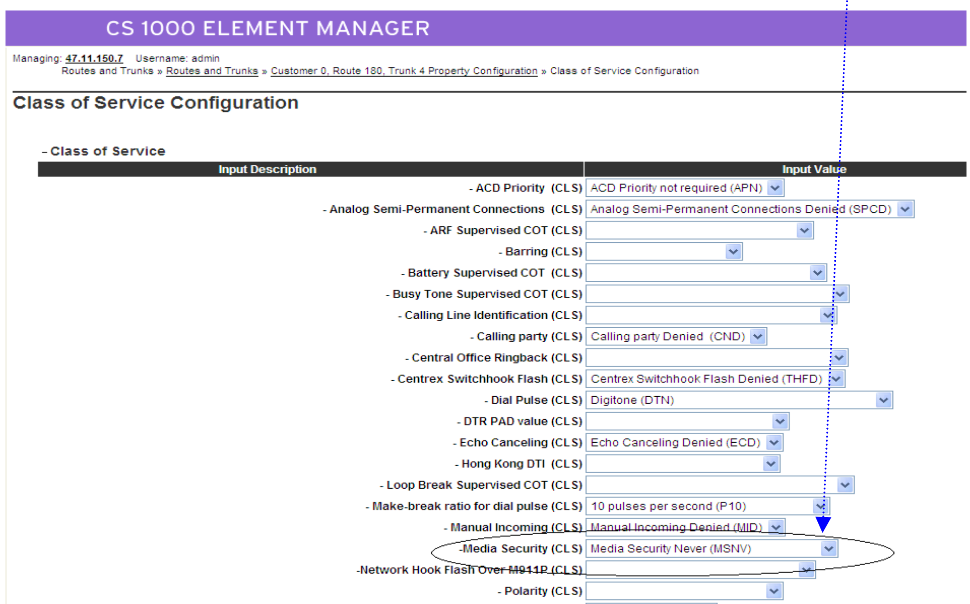
<u>IMPORTANT</u>: DO NOT enter domain names. Doing so will cause messaging will fail.



- continued on next page –

SRTP (Media Security) at Trunk Level (To AVAYA AURA MESSAGING) must be set to MSNV or disabled.

- From the UCM main menu, **click on** Routes and Trunks, then Customer, Route, Trunks to configure the Class of Service.
- Click Edit / Click Edit Class of Service.
- For SRTP: Media Security (CLS) should be set to *Media Security Never (MSNV)* using the drop down menu as circled below. Repeat this for all the trunks in the route.



<u>Note</u>: On the CS1000 you must configure all the stations/sets so they forward calls to the Avaya Aura Messaging Server. To do this you use LD11.

LD 11 – (Select the set type and associated TN number)

FDN: 2300 (AVAYA AURA MESSAGING Pilot Number)

Hunt: 2300 (AVAYA AURA MESSAGING Pilot Number)

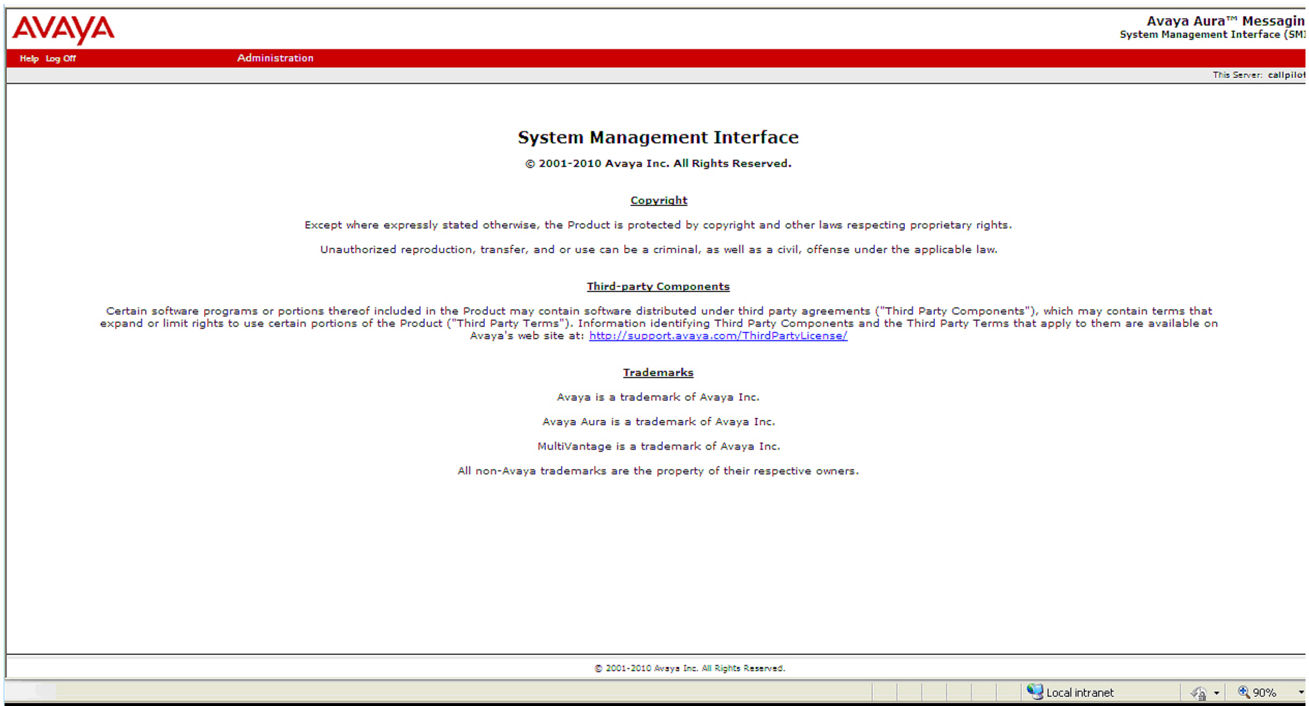
- continued on next page –

Configuring the Aura Messaging Server

6.0 Configuring the AURA Messaging Server

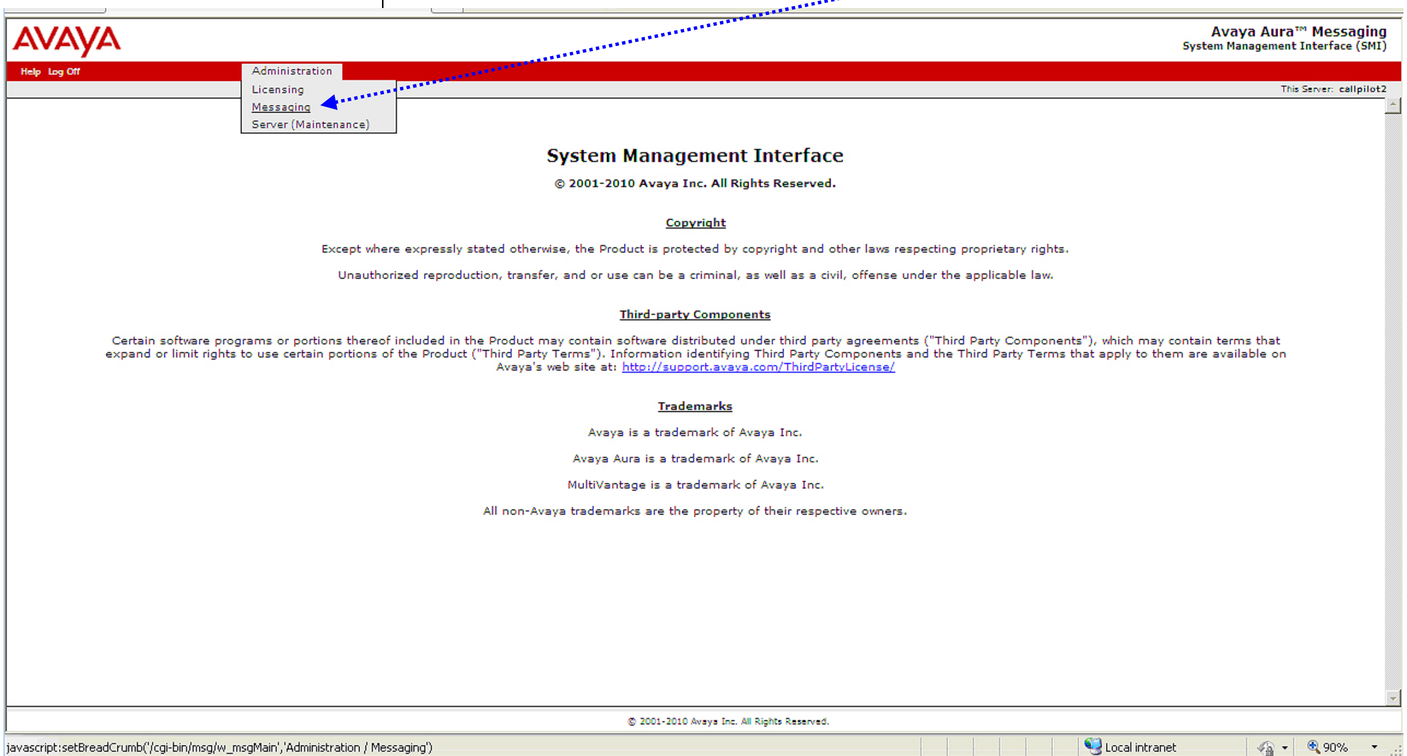
Configuring the AMS platform for proper PBX integration requires that the parameters are set as indicated in the screens below.

- When you first login to the Avaya Aura Messaging Server you will see the System Management Interface screen shown below.



- continued on next page –

- Chose the Administration pull-down and then chose Messaging



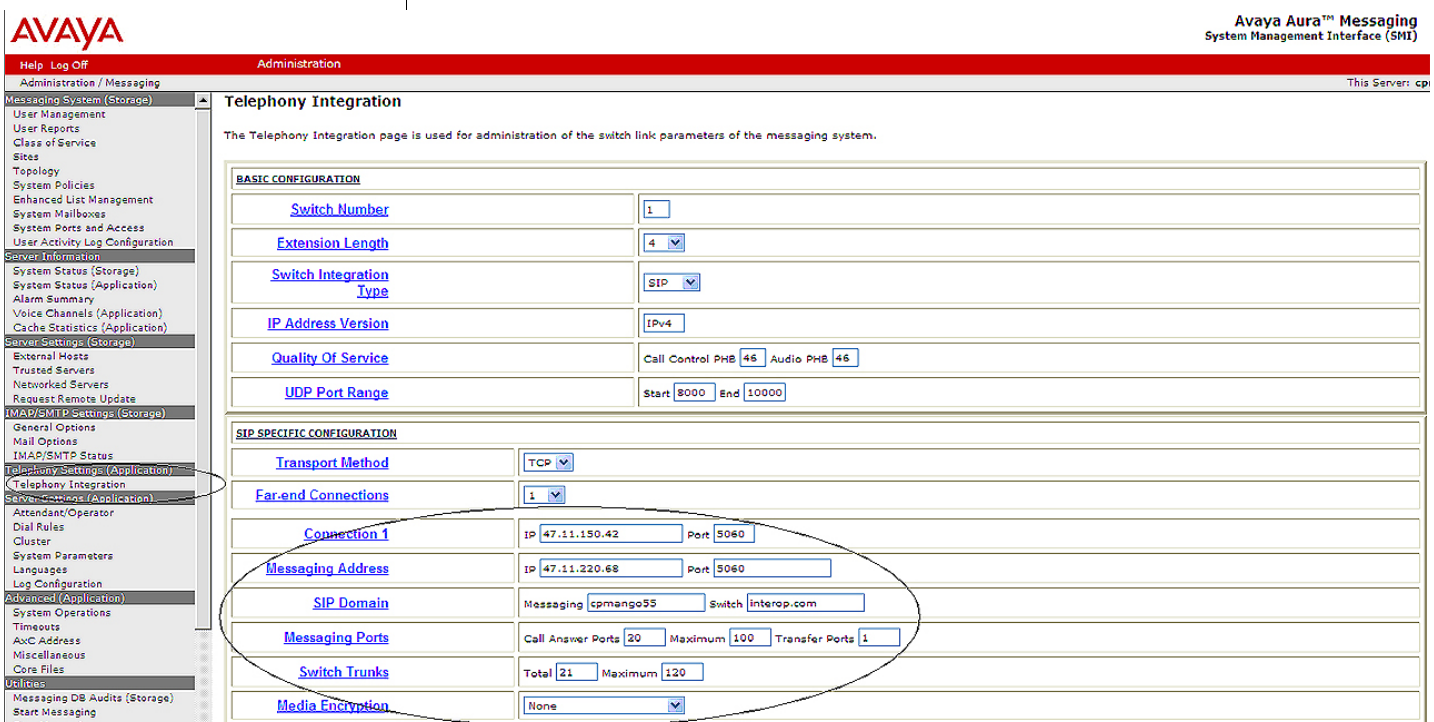
- continued on next page –

- The *Messaging Administration* screen below will be displayed.
- In the <u>left panel</u> scroll down until you see *Telephony Integration*, found under the group *Telephone Settings (Application)*, and click on it.

- continued on next page –

- The *Telephony Integration* screen will now be displayed. All the settings you need to administer the Avaya Aura Messaging Server for integration are done from this screen.

NOTE: The screen is divided into two sections: *BASIC CONFIGURATION* and *SIP SPECIFIC CONFIGURATION (circled)*. These sections are show separately in closer views on the pages that follow with details on the parameters needed to configure Avaya Aura Messaging for Direct Integration.



- continued on next page –

- The **BASIC CONFIGURATION** section of the Switch Link Administration Screen is shown below. Please note, the settings shown here are for example only. Your settings may be different, 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 = *4* *<extension length up to 10 digits>*. **Note**: The extension length of 4 applies to our sample configuration. Your extension length must match your dial plan.

IP Address Version = *Ipv4* (leave as default)

Switch Integration Type = *SIP*

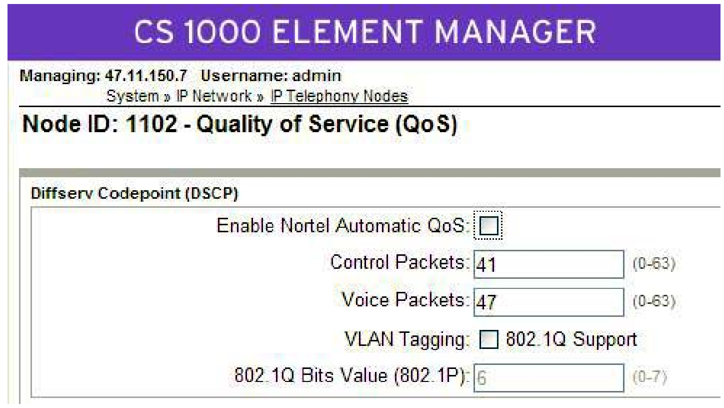
Quality of Service = *46 (*A Value of *0* to *63* may be used.)

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

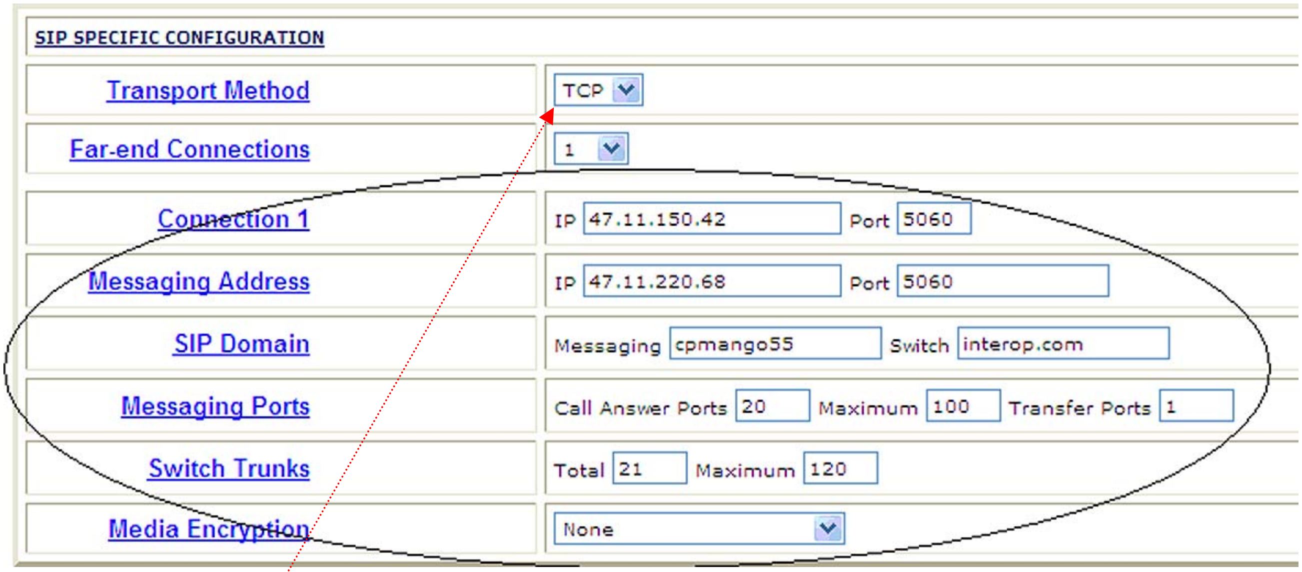
These values must match the corresponding numbers in the PBX to ensure good voice quality. Please see value in *Voice Packets* field in CS1000 QOS screen below

UDP Port Range: Start: **8000** (Specify a starting UDP port number for RTP. The **End** port is calculated automatically.

Please note this screen is found in the CS1000. It is provided to show where the user can locate the QOS values to match those in Avaya Aura Messaging.



The **SIP SPECIFIC CONFIGURATION** section of the Switch Link Administration Screen is shown below. Please note, the settings shown here are for example only. Your settings may be different, please refer to the configuration details shown below to administer your site.



SIP SPECIFIC CONFIGURATION	
Transport Method	TCP
Far-end Connections	1
Connection 1	IP 47.11.150.42 Port 5060
Messaging Address	IP 47.11.220.68 Port 5060
SIP Domain	Messaging cpmango55 Switch interop.com
Messaging Ports	Call Answer Ports 20 Maximum 100 Transfer Ports 1
Switch Trunks	Total 21 Maximum 120
Media Encryption	None

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.

***Note for TLS**

At the current time TLS is NOT supported.

Please use TCP for your *Transportation Method*.

Far-end Connections = *1*. This is the number of far-end connections to administer.

Connection 1 = ***47.11.150.42*** Enter the Signaling Server IP Gateway Node IP Address and Port (usually *5060* for *TCP* or *5061* for *TLS)*. **Note**: the address shown in the screen above is an example used for our sample configuration, your IP address will be different.

Messaging Address = ***47.11.222.68*** (this IP address field is always read only). Enter the Port number (usually *5060* for *TCP* or *5061* for *TLS)* for messaging. **Note**: the address shown in the screen above is an example used for our sample configuration, your IP address will be different.

SIP Domain = Messaging: *cpmango55;* Switch = interop.com **Note**: the Domain Names noted here are examples used for our sample configuration. The Messaging domain name here must match the Host Name of Avaya Aura Messaging; the Switch name must is the Domain Name on your PBX.

At the current time **Media Encryption** is NOT SUPPORTED.

Please set Media Encryption to None

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).
- *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

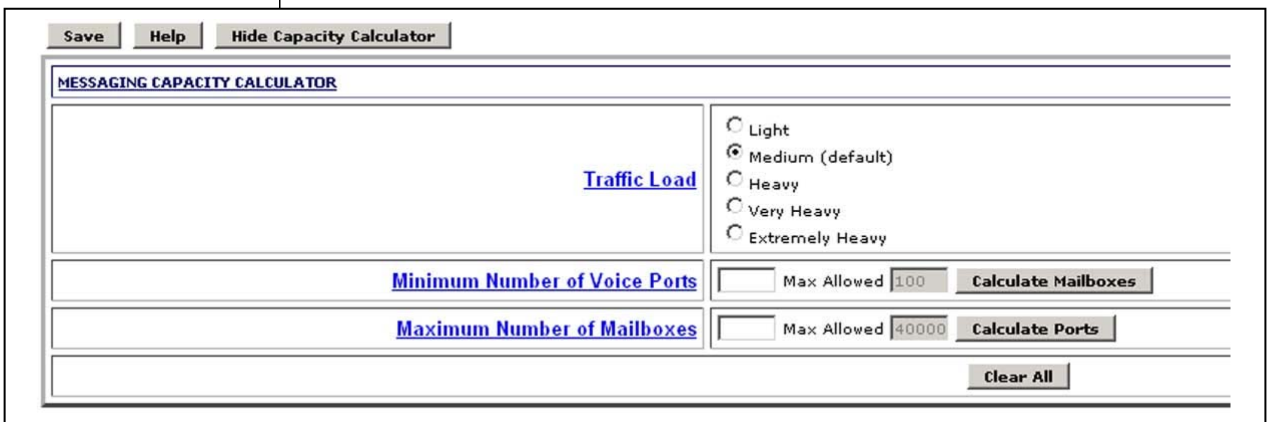
Chose the encryption (*one of the 3 choices noted above*) that matches what is administered in the IP Codec Set used for this integration.

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.

- continued on next page –

6.1 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.*

8.0 SPECIAL NOTES / CONSIDERATIONS / ALTERNATIVES

8.1 **SIP integrations may not be reliable for TTY** if the IP network is unable to support uncompressed audio with no packet loss. For this reason **Avaya does not support TTY 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 Avaya Aura Messaging currently does not support E.164 format numbering or any mailbox or extension number exceeding 10-digits.

8.4 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.

8.5 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".

- continued on next page –

CHANGE HISTORY		
Revision	**Issue Date**	**Reason for Change**
A	1/27/2012	Moving document to GA status. No content changes.
B	4/4/2012	Removed “DRAFT” watermark & clarified CS1K release under direct connect.
C	4/25/2012	Added a special note to section 8.5 to address multiple DNs (extensions) to one physical phone. Configuration clarification concern.
D	5/9/2012	Clarification under Section 8 regarding CODECs.
E	6/11/2012	Hardening support for CS1K6.0 DIRECT only.
F	6/25/2012	Removed DRAFT note.

AVAYA Inc.
4655 Great America Parkway
Santa Clara CA 95054
+1-866-Go-Avaya
From Outside the US: +1 (908) 953-6000
http://www.avaya.com