White Paper / Application Note

Configuring Avaya Communication Server 1000E and Avaya Aura when deploying Avaya Communicator for Microsoft Lync (SIP) - Issue 1.0

Abstract

This Application Note describes the detailed procedures for configuring Avaya Communication Server 1000 and Avaya Aura when deploying Avaya Communicator for Microsoft Lync. This solution consists of the Avaya Communication Server 1000 and the Avaya Aura® solution. In the sample configuration described herein, an Avaya Communicator for Lync user has their primary call control (SIP) on the Avaya Aura Communication Manager. Communicator for Lync uses its Other Phone Mode (Aka telecommuter Mode) to place and receive calls through their Avaya Communication Server 1000E desk phone, The user continues to utilize their single Avaya CallPilot voice mailbox

The steps documented in this Application Note focus on how these attributes are configured across the solution.

Information in these Application Notes has been obtained through Solution Verification full stack testing and additional technical discussions.

Table of Contents

1. Introduction

This Application Note describes the procedures for configuring Avaya Communication Server 1000E Release 7.6 and Avaya Aura® solution Release 6.4 with Avaya Communicator for Microsoft Lync release 6.4 (SIP end point). Screenshots in this document may vary slightly with the different release of the products but methods and procedures remains consistent.

This solution is based on the combined Avaya Communication Server 1000 / Aura Communication Manager solution (formally known as Collaboration pack). In this deployment Avaya Aura® Midsize Enterprise or Avaya Aura® with discreet server components is used to extend the Avaya Communicator for Microsoft Lync functionality to Communication Server 1000 (CS 1000) users.

This solution is suitable for Customer's who wish to retain their CS 1000 desk phone and use the Avaya Aura to add additional functionality such as Video support with Avaya Lync Integration.

With this solution Avaya Aura Communication Manger (CM) is used to provide call controlled over the users extension. This allows ACA to provide softphone support (Computer Mode) with Avaya Lync integration register directly to CM as a SIP end point and the user's Media (Voice and Video) presented to the user's PC. The user can make and receive calls using their CS 1000 desk phone by selecting Other Phone Mode. The user is able to retain the usage of their Call Pilot mail box.

The solution supports the following functionality:

- Utilize the Lync 2010/Lync 2013 client for IM and Presence, Microsoft Lync Voice and Video capability are turned off
- Make a Voice or Video[1] call from their Contact list, Outlook, Excel, Word, PowerPoint, or Internet Explorer using their Avaya infrastructure
- Select if they wish to use their computer or another device such as a CS 1000 set or mobile devices to make and receive calls
- Escalate from an IM to a Voice or Video[1] call
- Control active calls via a conversation window which provides mid call options such as end call, place the call on hold, insert DTMF digits into the call and escalate to Video[1]
- Receive a toast pop-up of an incoming call with information on who is calling and perform actions such as Answer the call
- Allows the user to Automatically Answer with video of have the manual option to accept or decline a Video[1] call.

Note 1: Video support is applicable to Computer Mode. A video call can't be instigated when using Other Phone Mode to control the user's CS 1000, Mobile or other device

This Application note aims at minimizing the impact to the CS 1000 User configuration and introducing call routing which ensures calls can be correctly presented to the appropriate end point.

This document makes the following assumptions

1. The enterprise has an Active directory populated with E164 Numbers which are synchronized with the Lync Address book
2. The CS 1000 users existing extensions number is are a sub set of these E164 numbers
3. Their new Aura extensions is also be an subset of the E164 numbers

Example: User’s full E164 extension is +1 303 447xxxx, their existing CS 1000 extension was 7xxxx, and their new Communication Manager extension will be 447xxxx.

CS 1000 clients tested in the sample configuration covered by this application note include the 11xx and i2002p2, i2004p2 UNIStim IP desk phones and 39xx digital desk phones. Although not explicitly tested in this environment Avaya 12xx sets are also supported.

Avaya Aura® Midsize Enterprise R6.3 currently supports up to 2,000 users on a single server platform that includes virtualized instances of Avaya Aura® Session Manager, Avaya Aura® System Manager, Avaya Aura® Communication Manager and Avaya Aura® Presence Services. A G430 or G450 gateway is also included as standard with the Avaya Aura® Midsize Enterprise.

In this deployment Avaya Aura® Presence Services is not utilized.

Where a higher user capacity is required or where the Customer has already deployed discreet elements of the Aura solution then components Session Manager, System Managers and Communication Manager can be deployed/reused.

The Avaya Communication Server 1000E R7.6 provides advanced telephony capability via M3900 series digital desk phones and 1100 / 1200 series IP desk phones with UNIStim software. Connectivity with the Avaya Aura® system is via SIP trunks and PSTN connectivity is provided through ISDN trunks on the CS 1000.

Avaya CallPilot® provides a centralized voice mail capability for all users in the solution offering a centralized voice mailbox and associated Message Waiting Indication (MWI) functionality. Network Message Service (NMS) capability must been abled on CallPilot to allow the transmission of MWI to clients on the Avaya Aura® Midsize Enterprise system.

Dial-in and meet-me conference services only are provided with an optional Avaya Aura® Conference server.

This Application Note will document the steps necessary to configure the main components of the Avaya CS 1000 and Avaya Aura to accommodate one Communicator for Lync user. The document is based on an existing sample configuration used in the testing of the solution.

Steps described in this document include:

1. Configure Avaya Communication Server 1000E
2. Configure Avaya Aura® Session Manager,
3. Configure Avaya Aura® Communication Manager,
4. User Management– configure Communicator for Lync user,
5. Verification Steps.

Detailed administration of other aspects of the CS 1000/Aura or additional equipment to support the installation (e.g. Active Directory / Domain Name Servers, Voice / Data Network equipment, Wireless LAN infrastructure, etc.) will not be described as it is outside the scope of this Application Note.

Administration of CallPilot for this solution is not covered in this application note as are the administration of optional CS 1000 / Aura solution components such as Avaya Aura® Conference (AAC), ASBCE for Remote User interactions, etc. References to relevant documentation sources are provided in **Section 10** to cover these.

Throughout this Application Note, the term “Avaya Communication Server 1000 / Aura” may sometimes be abbreviated and referred as “CS 1000/Aura” and the product name “Avaya Communicator for Microsoft Lync” will be shortened to “Avaya Communicator for Lync” or “Communicator for Lync”. Session Manager will be abbreviated to SM and Communication Manager to CM

2. Interoperability Testing

A reference configuration containing all of the equipment for the CS 1000 / Aura was installed and a large number of tests cases were executed to ensure functionality of the various user endpoints supported and interoperability between CS 1000 and Avaya Aura® Mid-Size Enterprise solution.

In this configuration Avaya Communicator for Microsoft Lync user is configured on Communication Manager (CM) and registered to Session manager as SIP end point and as such call control is anchored on CM. The user has the ability to make and receive call using their Other Phone (with their CS 1000 phone) or Computer Mode.

When, the user selects Other Phone Mode and makes a call, CM will call the user’s desk phone 2xxxx and when answered will proceed to call the called party. Communicator for Lync will present the user with a Conversation window which will allow them to control the call from their PC.

For an incoming call, it will be presented to CS 1000 virtual extension 7xxxx and using PCA this call will be forwarded to SM/CM extension 447xxxx. Communicator for Lync will present a Toast Pop up notifying the user to answer their call on their desk phone. When answered, Communicator for Lync will present the user with a Conversation window which will allow them to control the call from their PC.

When in Computer mode and the user make a call, CM will place the call directly to the called party. Communicator for Lync will present the user with a Conversation window which will allow them to control the call from their PC. For an incoming call it will be presented to CS 1000 virtual extension 7xxxx and using PCA this call will be forwarded to SM/CM extension 447xxxx. Communicator for Lync will present a Toast Pop up notifying to the user of the incoming call. The user can answer/decline this call by selecting the accept/ignore call option on the toast pop up. When answered, the user will be presented with a Conversation window which will allow them to control the call from their PC.

1. Communicator for Lync user’s full E.164 number: +1 303 447 xxxx
    a. CS 1000desk phone DN: 2xxxx
    b. PCA virtual DN: 7xxxx
    c. CM extension of Lync: 447xxxx
2. CS 1000 user with additional end point on Aura i.e. Avaya Communicator for Windows or Avaya Communicator for iPhone/Android clients
    a. User’s full E.164 number: +1 303 447 xxxx
    b. PCA virtual DN: 7xxxx
    c. CM extension of Lync: 447xxxx
3. Native users on CS 1000
    a. User’s full E.164 number: +1 303 447 xxxx
    b. CS 1000desk phone DN: 7xxxx.

When the Communicator for Lync user want to make a call to another CS 1000 user they can select the user from their Lync Contact list, Search Lync for this contact or enter the user CS 1000 DN. Communicator for Lync will take this number, be it the 5 digit CS 1000 extension or the users full E164 number and convert this to the CS 1000 5 digit extension i.e. 7xxxx. This number is then passed to CM which routes the call via Session Manager to the CS 1000. If this call is destined for a native CS 1000 user then CS 1000 will present the call to their device. If the call is for another Communicator for Lync user then the call will receive its PCA treatment and be routed back to Aura.

It is recommended that for correct CLID, calls are made using the Communicator for Lync client. Load 15 can be used to configure the local desk phones CLID to display the users E164 number.

In the case of the Lync Client being logged out, call no answer or call busy, call redirection will occur which will redirection the call to the user's desk phone and / or to the user's single CallPilot voice mailbox.

PSTN calling is achieved with ISDN trunks off the CS 1000 for all clients.

It is presumed that the CS 1000E Release 7.6 system and software and the CallPilot integrated server and software have already been provisioned. This combination represents an existing CS 1000 Customer configuration. In addition, the extra components of the CS 1000 / Aura solution such as Avaya Aura® Midsize Enterprise system, various SIP endpoints and network infrastructure (LAN, WLAN, an optional Session Border Controller, etc.) and an optional Avaya Aura® Conferencing server are also presumed to have been provisioned.

This Application Note documents the procedures necessary to configure the main components of the solution (CS 1000, Session Manager and Communication Manager) including the SIP trunking between the CS 1000 and Session Manager and subsequent user administration for one new Avaya Communicator for Microsoft Lync user.

Note: An incoming PSTN or CS 1000 originated call presented to the Communicator for Lync user in Other Phone Mode will be route back to the users CS 1000 desk phone. Since this call continued to hosted on Aura Communication manager two trunks are consumed between the CS 1000 and Aura systems

2.1. Test Description and Coverage

To verify the interoperability and operation for Avaya Communicator for Microsoft Lync Users the following features and functionality were covered during the testing:

- Single E164 number for CLID on incoming and outgoing calls
- Single voice mailbox via CallPilot with associated MWI functionality
- Voicemail navigation for inbound and outbound calls via CallPilot
- Presence status during basic call scenarios with Lync as the presence aggregator
- Incoming and outgoing PSTN calls
- Point to point calls (call flows between each user / endpoint)

- Multiple call scenarios
- User features such as hold and resume (with music on hold)
- Call transfer and conference calls (Communicator for Lync Computer mode only)
- Caller ID presentation during incoming and call transfers
- Proper codec negotiations (G711 / G729 / direct and in-direct media)
- CS 1000 Attendant console interactions / call flows

The following was not tested with this solution:

- Interactions with ACE based Lync Integration (not supported)
- Any clients on the Aura client other than those explicitly referenced in this document are considered out of scope
- Third party CDR applications
- AAC multiple / cascading media servers
- AAC Ad-hoc conference and Dial-out not supported for the solution at this stage
- No requirement to test SIP clients registered to CS 1000 SIP Line Gateway
- Support of a peer CS 1000 system with the same ME (CS 1000 Networking)
- Geo-redundancy and branch solutions
- PSTN calls routed directly to Aura
- SIP Trunks to Service Provider
- Off-Net Call forwarding
- PRI or SIP trunks between CS 1000 and Communication Manager for convergence
- FAX call testing
- Traffic / Load testing
- Scale testing with endpoints
- Internationalization/ Localization testing
- Web Alive
- Identity Engine Analytics
- DTLS (secure signaling link) for UNIStim IP sets on CS 1000
- Network Address Translation (NAT) remote endpoints through ASBCE
- Callpilot Desktop Messaging
- ACD agent as a interactions with Avaya Communicator for Microsoft Lync users
- CS 1000 features which cannot be supported with Communication Manager, e.g. MADN (Multiple Appearance DN), Call Park, Call Pickup, Boss Secretary, etc

2.2. Test Results and Observations

Interoperability testing of Avaya Communicator for Microsoft Lync user with CS 1000 / Aura solution was completed successfully with the following observations made and issues / limitations noted.

- No Avaya Communicator for Microsoft Lync Conversation window will be displayed if calls are placed directly from the user's desk phone. Please use the Lync client to originate calls if you wish to maintain call control through your PC
- Similarly incoming calls must be answered by selecting "Accept" on the call notification pop up. If the user answers the call directly on their desk phone, a conversation window won't be displayed
- Placing calls on Hold or activating Call Transfer or Conferencing capabilities on the Users CS 1000 desk phone will not be reflected on the Communicator for Lync call window

3. Reference Configuration

The following diagram (**Figure 1**) shows the reference configuration used in the testing of a CS 1000 / Aura sample solution. It depicts a possible CS 1000 /Aura solution configuration.

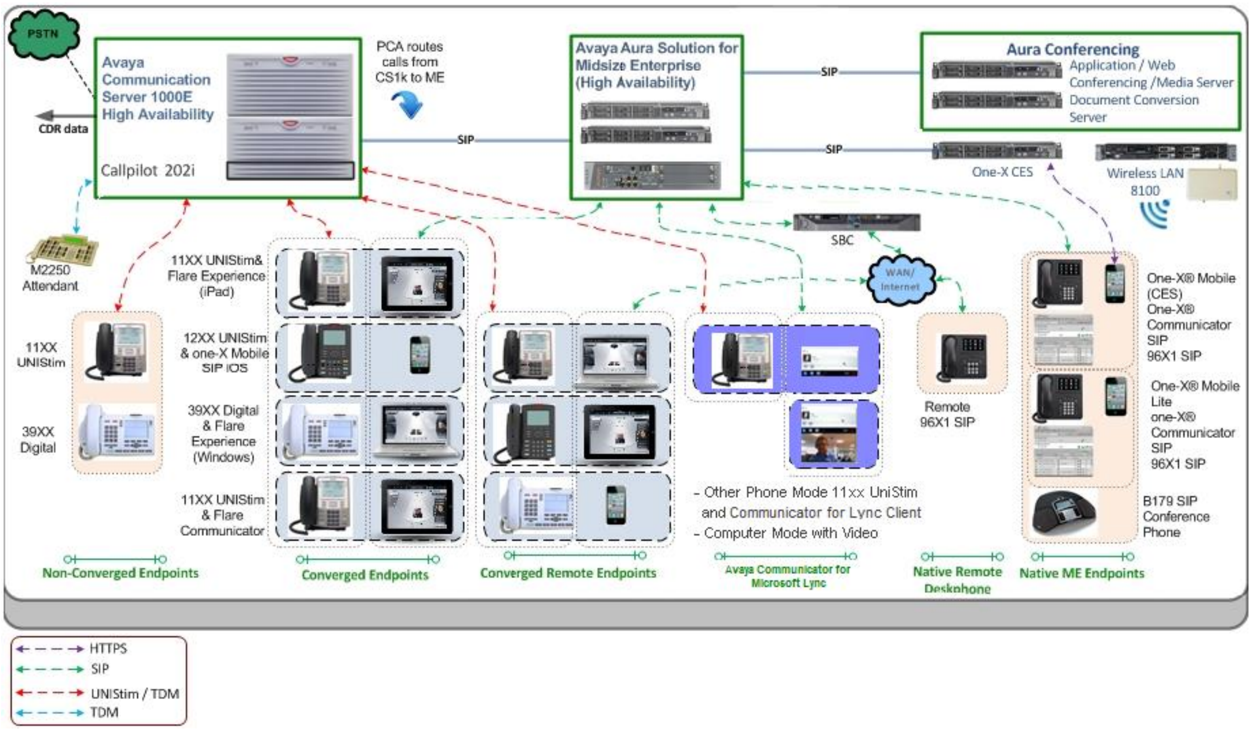


Figure 1: Typical CS 1000 / Aura Solution Configuration

Four groups of users are defined as follows for CS 1000 / Aura solution:

Group 1 - Non Converged User: An existing CS 1000 user that continues to use their CS 1000 desk phone with no association to any client on the Aura. A CallPilot voice mailbox is assumed to be already in place for a non-converged user. Supported endpoints are:

- 11xx IP (UNISTIM) desk phone
- 12xx IP (UNISTIM) desk phone
- I2002p2/i2004p2 (UNISTIM) desk phone
- 39xx Digital desk phone

Group 2 - Converged User: An existing CS 1000 user that continues to use their CS 1000 desk phone and is also associated, via PCA on the CS 1000 side, with a SIP client on the Aura. The endpoints in this arrangement are referred to as “Converged Endpoints”. Some of these Aura clients may be remote users who are connected into the Enterprise network over the WAN via ASBCE. In that case, the endpoints can be referred to as “Converged Remote Endpoints”. Supported endpoints in this group are:

- 11xx IP (UNIStim) desk phone (on CS 1000)
- 12xx IP (UNIStim) desk phone (on CS 1000)

- I2002p2/i2004p2 (UNISTIM) desk phone
- 39xx Digital desk phone (on CS 1000)
- one-X Mobile (SIP) for iOS (on ME)
- Avaya Communicator for iPhone (on ME)
- Avaya Communicator for Android (on ME)
- Avaya Communicator for Windows (on ME)
- Remote Avaya Communicator clients registered to ME via ASBCE

Group 3 – Avaya Communicator for Microsoft Lync User: A CS 1000 user who Call control moves from CS 1000 to CM. This user's previous extension is converted to a Virtual DN with PCA to route the call via SM to CM. Their CS 1000 desk phone is configured with a new extension number. Thus user also has Communicator for Lync client registered to SM as a SIP end point and uses Other Phone Mode (aka telecommuter mode) to make and receive call trough their new CS 1000 Desk phone DN. A CallPilot voice mailbox is assumed to be already in place for this user but has to be updated to reflect the new configuration. Supported CS 1000 endpoints are:

- 11xx IP (UNISTIM) desk phone
- 12xx IP (UNISTIM) desk phone
- I2002p2/i2004p2 (UNISTIM) desk phone
- 39xx Digital desk phone
- Analogue Devices

Note: Other Phone Mode can also be used to make and receive calls with mobile and other devices such as hotel room and home phone which have a defined E164 number.

Group 4 - Native User: A user on the Aura that has no corresponding CS 1000 desk phone. Services for these users are provided via clients on the Aura only. Incoming PSTN calls or calls from non-converged users are routed to the Aura over SIP trunk from CS 1000via PCA. The endpoints in this arrangement are referred to as "Native Endpoints". Some of these Aura clients may also be remote users connected into the Enterprise network over the WAN via ASBCE. In that case, the endpoints can be referred to as "Native Remote Endpoints". Native clients supported are:

- Avaya B179 SIP Conference phone
- Avaya one-X Communicator (SIP) with Audio provided by an Avaya one-X® Desk phone96x1 SIP
- Avaya one-X Communicator (SIP) with Audio provided by an Avaya one-X® Desk phone96x1 SIP phoneand extended with EC500 to an Avaya one-X® Mobile Lite.
- Avaya one-X® Mobile with Client Enablement Services (CES) which is supported with an Avaya one-X® Desk phone 96x1 SIP on ME only (i.e. not supported for a CS 1000 phone).
- Avaya one-X® Desk phone 96x1 SIP
- Avaya Communicator for iPhone (on ME)
- Avaya Communicator for Android (on ME)
- Avaya Communicator for Windows (on ME)
- Remote Avaya Communicator clients registered to ME via ASBCE

- Remote Avaya one-X® Desk phone SIP on 96x1 registered to ME via ASBCE

Other components of the solution are also shown in **Figure 1** above such as AAC, One-X CES and Wireless LAN 8100 infrastructure and a description of how these are configured for the CS 1000 / Aura solution is outside of the scope of this application note. **Section 10** has references to other documentation guides and application notes relevant to the CS 1000 / Aura solution covering topics such as configuration of CallPilot voicemail for Communicator for Lync users in a CS 1000 / Aura solution.

This Application Note describes the configuration of the various system components required to enable an existing CS 1000 user to be to Communicator for Lync user. It does not covers the configuration required for provisioning converged or native users on the ME system.

For the purposes of this application note, a more simplified diagram of the CS 1000/ Aura configuration is shown below in **Figure 2**. This diagram serves as the basis for the configuration steps which will be described throughout this application note.

The numbering plan adopted for a Communicator for Lync user in the sample configuration has been designed such that a DN of the user on the Aura side and the DN of its equivalent on the CS 1000 side are both sub sets of the User E164 number. In this example, the users existing DN 70408 will be configured as a virtual extension, their 1140 UniStim telephone set has been provisioned on the CS 1000 system with a DN = 20408 and an Communicator for Lync user endpoint on the Avaya Aura® Midsize Enterprise system has been provisioned with a DN = 4470408. In Active Directory their work number is defined as User E164 number +1 303 447 0408 this is synchronized with the Lync Address book and appears as their Lync work number. The number routing and dial plan manipulations to allow the call routing of calls presented to the CS 1000 and / or Avaya Aura® Midsize Enterprise systems. The overall result is that any call made to user CS 1000 DN 70408 will automatically get routed to their new CM DN 4470408 and the user can adopt a mode to answer the call. By pre-selection Computer mode they are able to answer on their PC or pre-selection Other Phone mode to have calls presented to their Desk phone DN 20408.

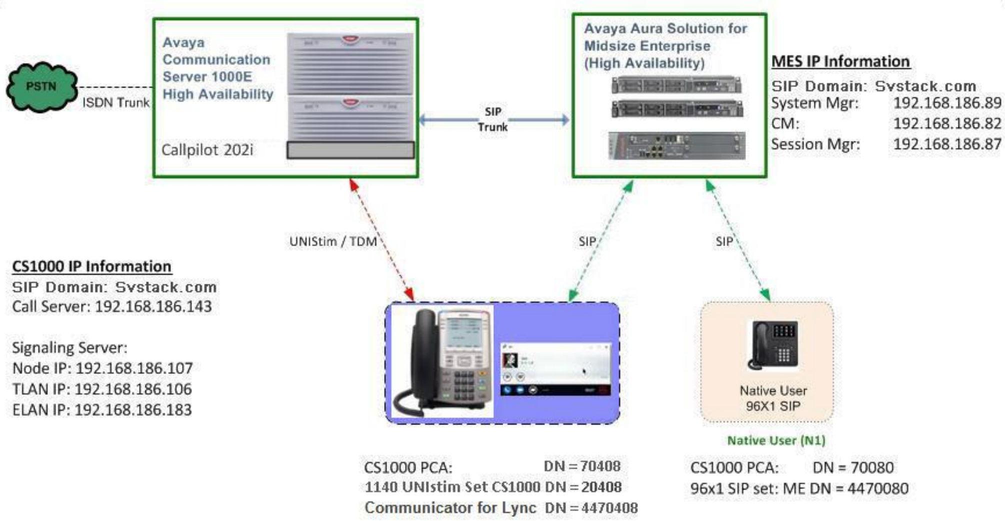


Figure 2: Simplified CS 1000 / Aura Configuration

4. Equipment and Software Validated

The following equipment and software were used for the sample CS 1000 /Aura solution test configuration.

Equipment/Software	Release/Version
Avaya Communications Server 1000E PBX	R7.6 + latest patches. i.e. 7.65
Avaya CallPilot running on a 202i integrated server	R5.01.01 + PEPs CP0501SU001S, CP501S01G08S, CP501S01G09C
Avaya 1100 Series IP Telephonesfor Avaya Communications Server 1000E	Firmware version 5.5 (UNIStim)
Avaya 1200 Series IP Telephonesfor Avaya Communications Server 1000E	Firmware version 5.5 (UNIStim)
Avaya 3900 Series TDM Telephonesfor Avaya Communications Server 1000E	Firmware version AA94 delivered with CS 1000 R7.6
Avaya Aura® Solution including Midsize Enterprise (ME)	Avaya Aura® Communication Manager 6.3.8/6.3.9/6.3.10 Avaya Aura® System Manager 6.3.8/6.3.9/6.3.10/6.3.11 Avaya Aura® Session Manager 6.3.8/6.3.9/6.3.10/6.3.11
Avaya Communicator for Microsoft Lync	6.4

5. Configure Avaya Communication Server 1000E

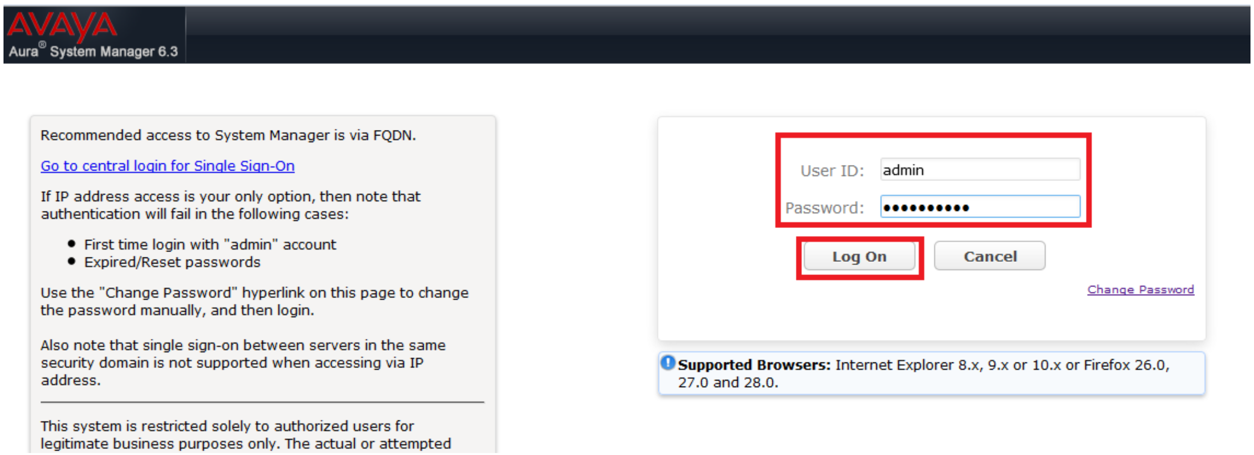
This section describes the details for configuring CS 1000E to route calls to the Communication Manager via Session Manager over a SIP trunk. These instructions assume that the CS 1000E has been registered as a member of the System Manager Security framework. In addition, these instructions also assume that the configuration of the CS 1000E Call Server and Signaling Server applications has been completed to support SIP trunks, IP (UNIStim) telephones and Digital telephones. Refer to **Section 10** for more information on how to administer these functions.

Using the Avaya Unified Communications Management (UCM) interface, the following administration steps will be described:

- Logon to Avaya Aura® System Manager
- Enable Avaya Unified Communications Manager services in Avaya Aura® System Manager
- Confirm Node and IP addresses
- Configure SIP Trunk to Avaya Aura® Session Manager
- Confirm Virtual D-Channel, Routes and Trunks
  - Confirm Virtual D-Channel Configuration
  - Confirm SIP Route and Trunk Configuration
- Configure ESN for Route List Index and Digit Manipulation
  - Create Route List Index
  - Create Distant Steering Code

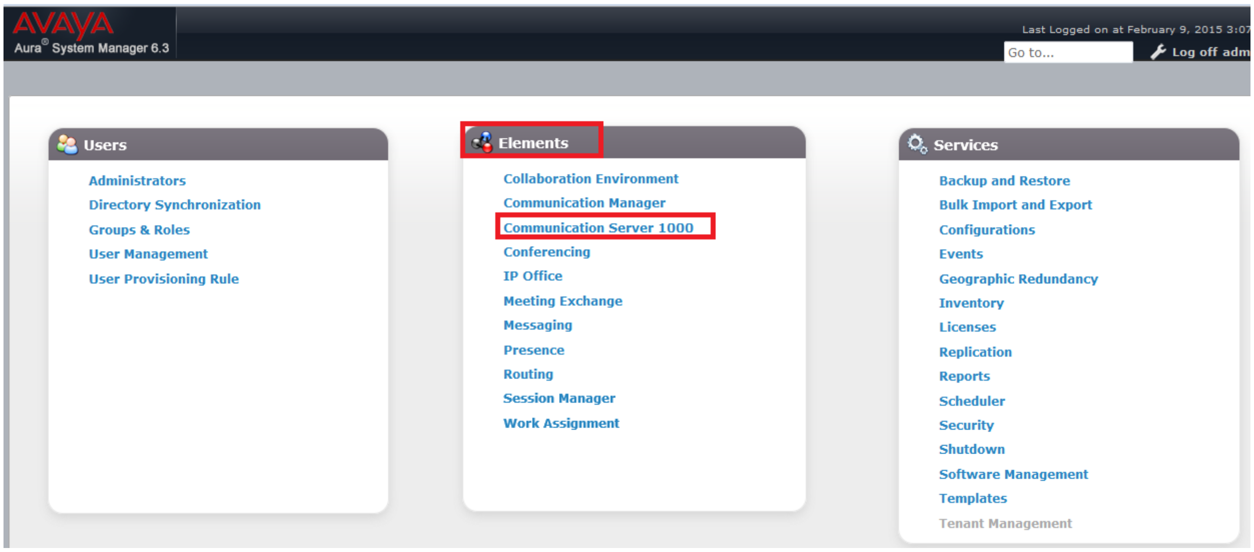
5.1. Logon to Avaya Aura® System Manager

Using any supported browser, access the web based GUI of System Manager by using the URL **https://<FQDN>/SMGR**, where **<FQDN>**is the Fully Qualified Domain Name of System Manager (for example https://messmgr.svstack.com/SMGR). In the **User ID** box, enter **admin** and enter the corresponding admin account password in the **Password** box. Click on the **Log On** button to login to the System Manager 6.3 console.

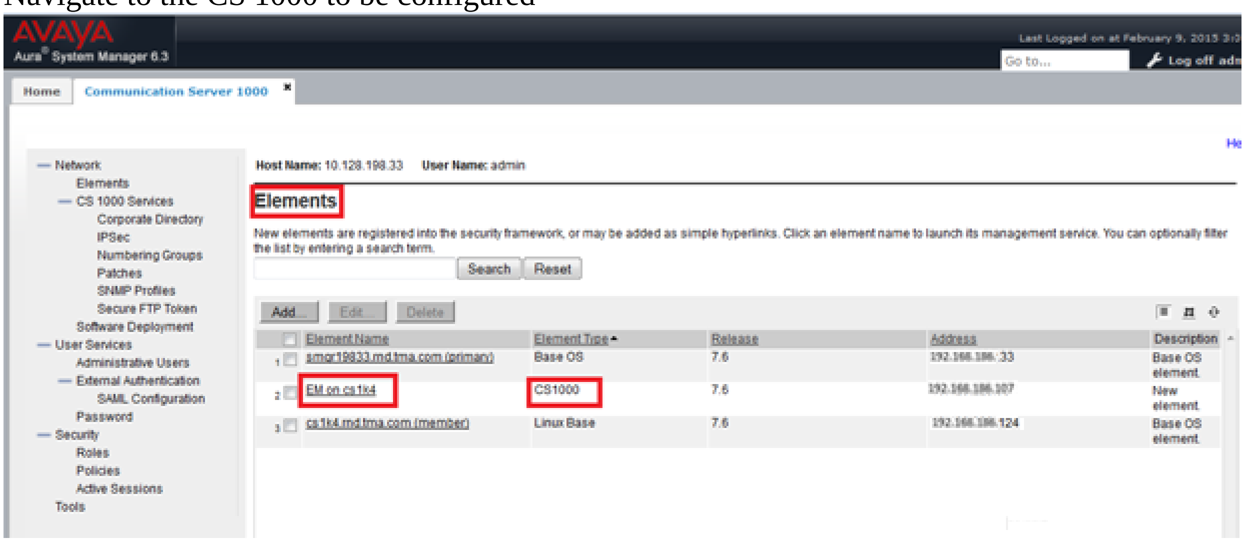


5.2. Navigate to CS 1000 Element Manager

To configure CS 1000 select Communication Server 1000 under the Elements

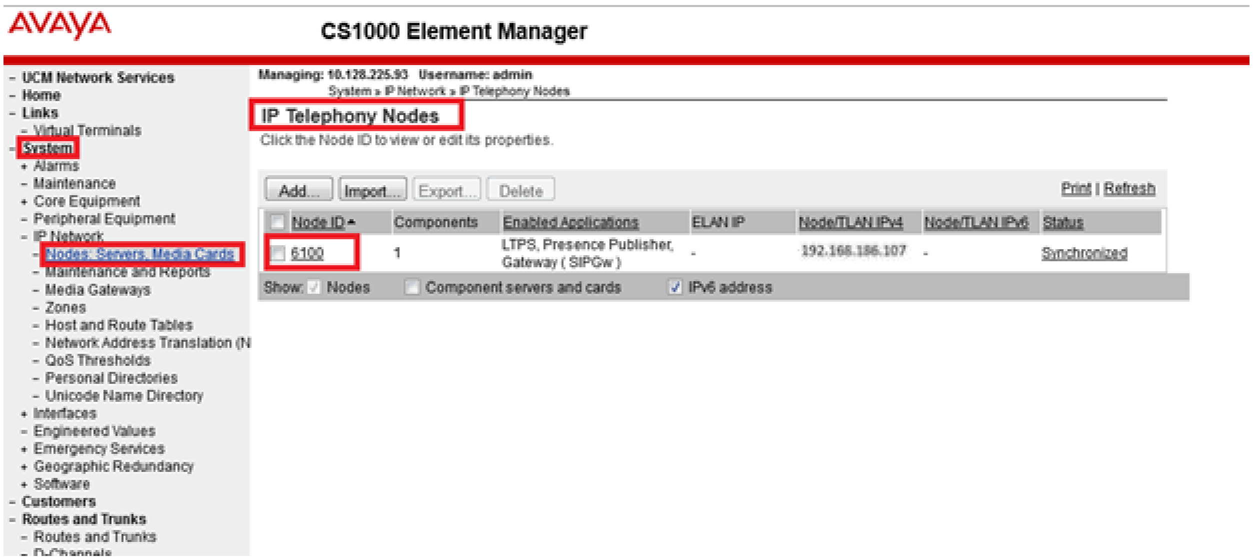


Navigate to the CS 1000 to be configured

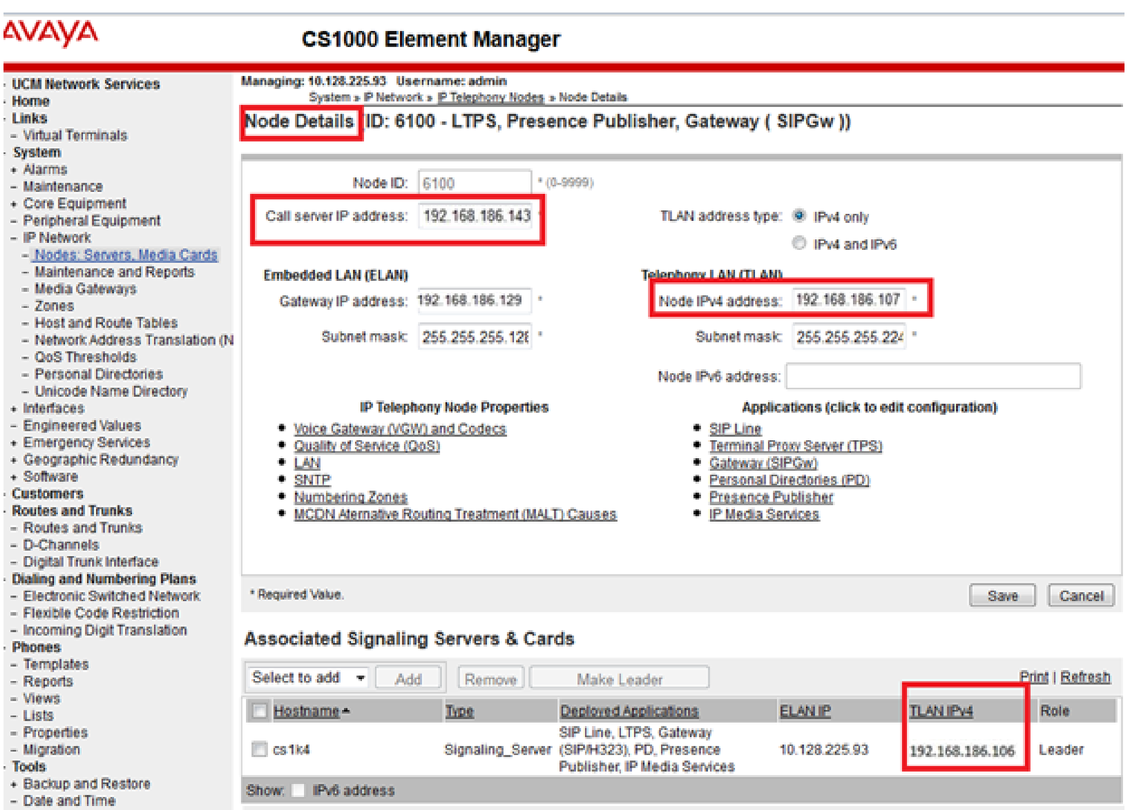


5.3. Confirm Node and IP Addresses

On the left hand side of the **CS 1000 Element Manager**, if not already expanded, expand the **System** list. Then expand the **IP Network** list and select **Nodes: Servers, Media Cards**. The **IP Telephony Nodes** page is displayed as shown below. Click on the Node ID number in the **Node ID** column to view details of the node (e.g. Node ID **6100**).

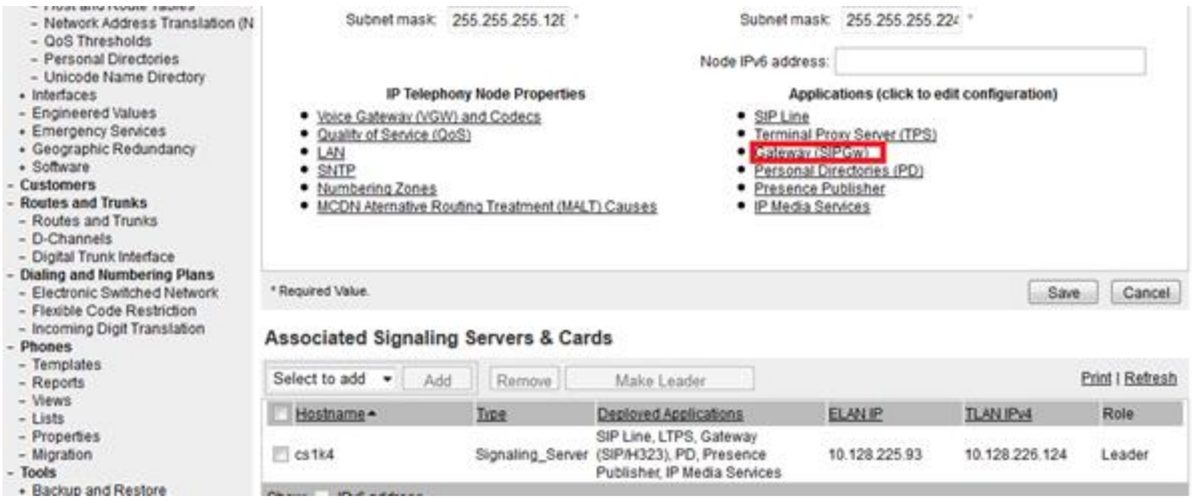


The **Node Details** screen is displayed with additional details as shown below. Make a note of The **Node IPv4 address, Call server IP address** and **TLAN IPv4** addresses of any Signaling Servers in the node. These addresses are used to configure other items later in this document.

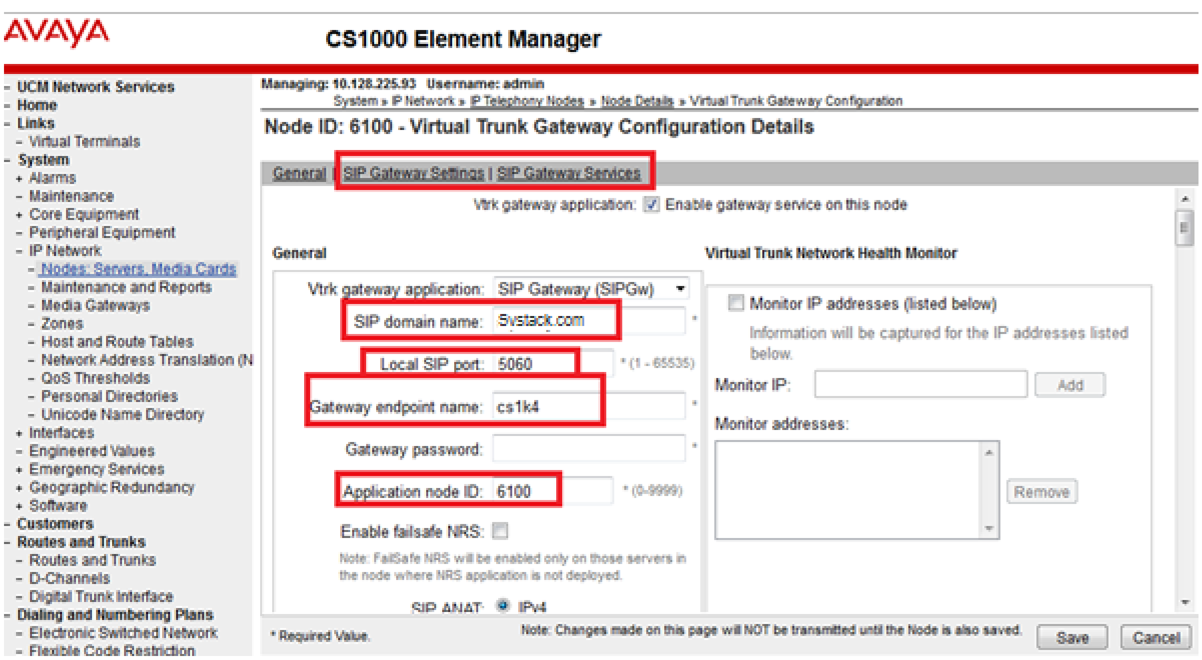


5.4. Configure SIP Trunk to Avaya Aura® Session Manager

While still in the **Node Details** screen as shown above in the previous **section 5.4**, use the scroll bar on the right side of the screen to navigate down to the **Applications** section and select the **Gateway (SIPGw)** link.

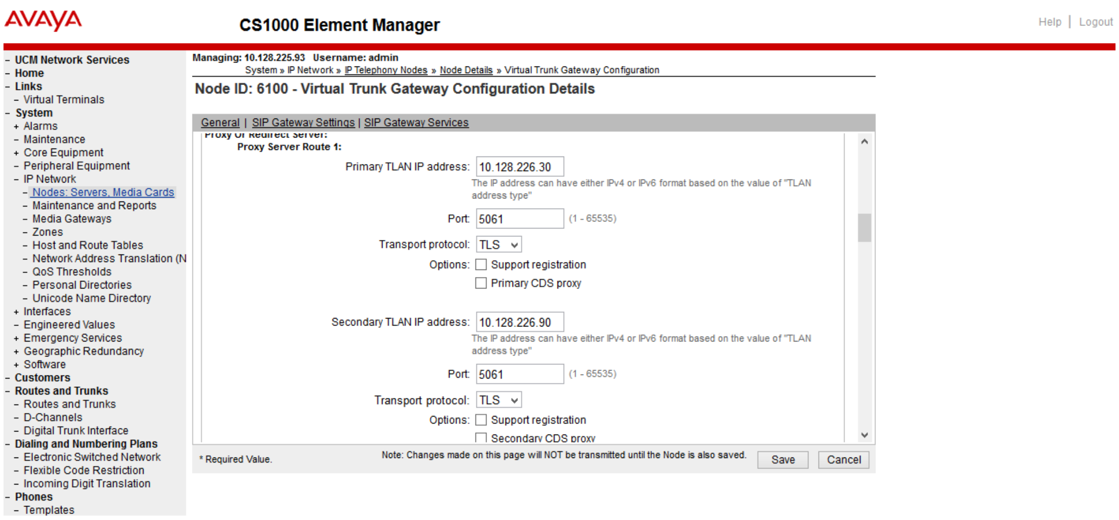


The **Node ID: 6100 - Virtual Trunk Gateway Configuration Details** page appears. Verify that the following fields have been pre-configured or enter new values if not (use default values for fields not specified here). The **SIP domain name** field should contain the SIP domain name for the solution (e.g. **Svstack.com**). Ensure that **5060** is entered in the **Local SIP port** field. A descriptive name should be entered in the **Gateway endpoint name** field (e.g. **cs1k4**). For the **Application node ID** field enter the Node ID value (e.g. **6100**). Then click on the **SIP Gateway Settings** link at the top of this page to jump to that section.

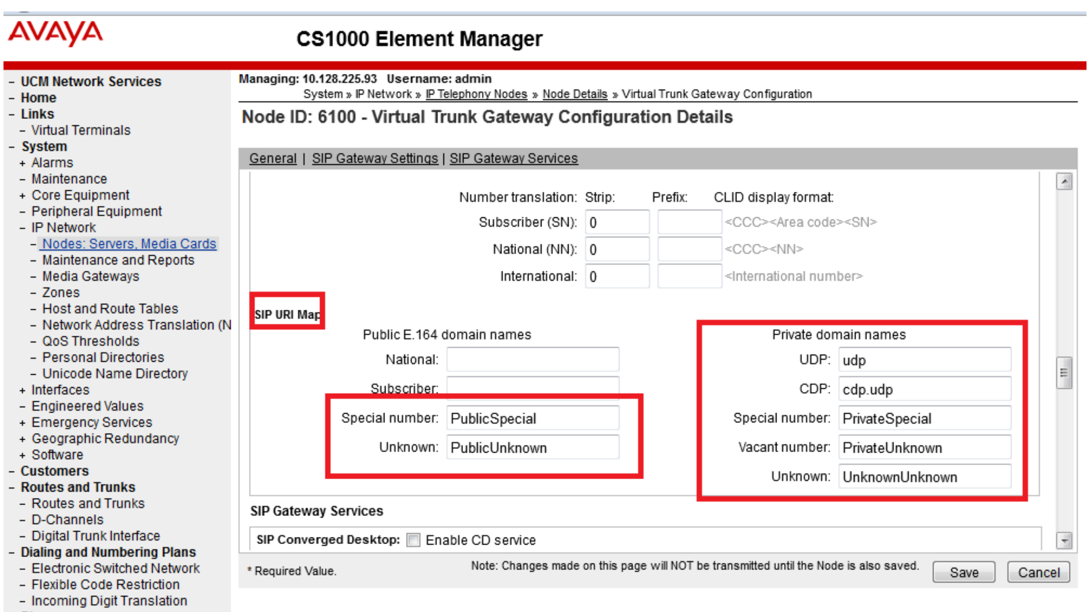


The SIP Gateway Settings page appears. Scroll down to the **Proxy or Redirect Server** section of the page. In the sub-section titled **Proxy Server Route 1**, enter the IP address of the Session Manager SIP signaling asset in the **Primary TLAN IP address** field (e.g. **192.168.186.87**). Enter **5061** in the **Port** field and select **TLS** as the **Transport protocol**.

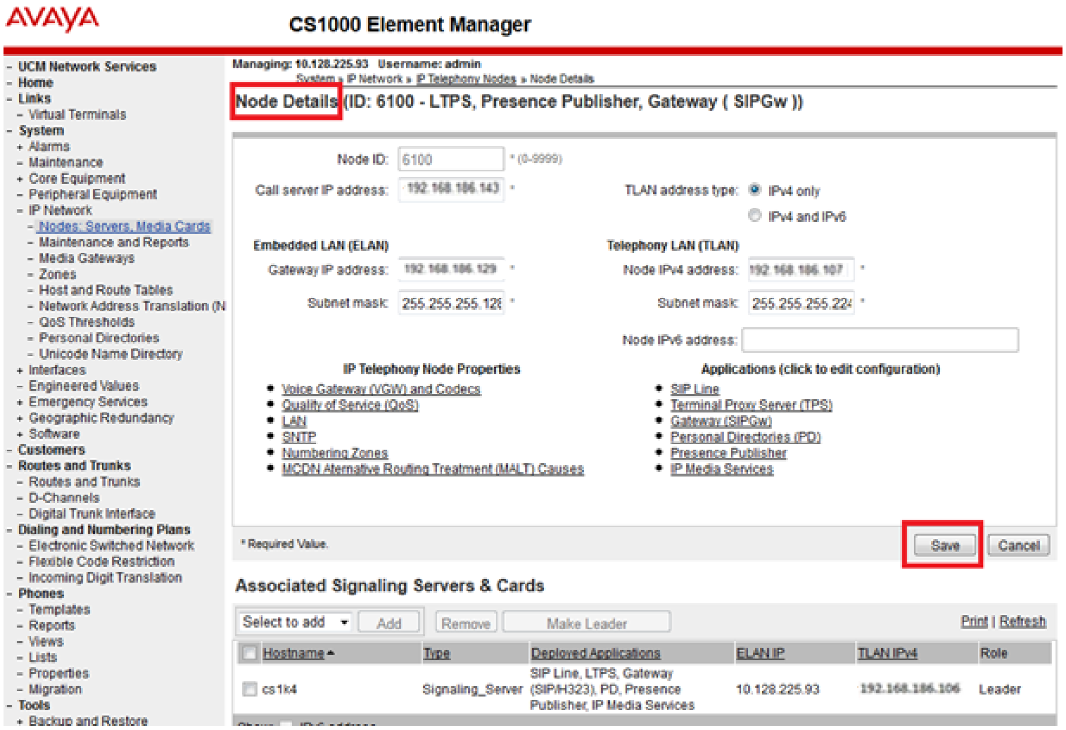
Note: For more information on configuring the system to use TLS, see the Application Note references in **Section 10**.



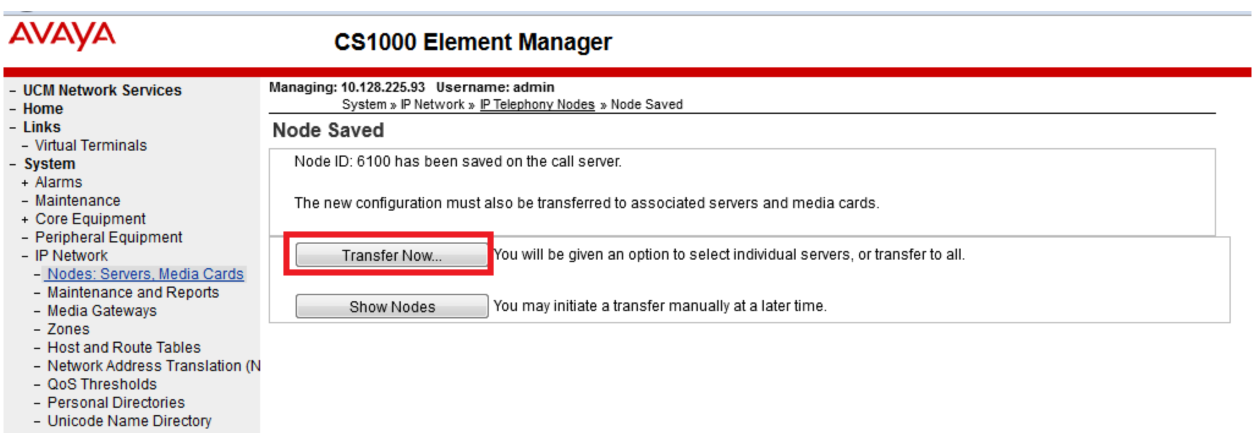
Scroll further down to the **SIP URI Map** section of the page. In the **Public E.164 domain names** and **Private domain names** sections, enter appropriate values in the fields to match the Customer specific solution. In the sample configuration, the values shown were used (**Note:** in the **Private domain names** section, the default entry for **Unknown** is “Unknown”. During test the default value was changed to “**UnknownTest**” to monitor the behavior during call flows). Click **Save** at the bottom of the screen to save any changes.



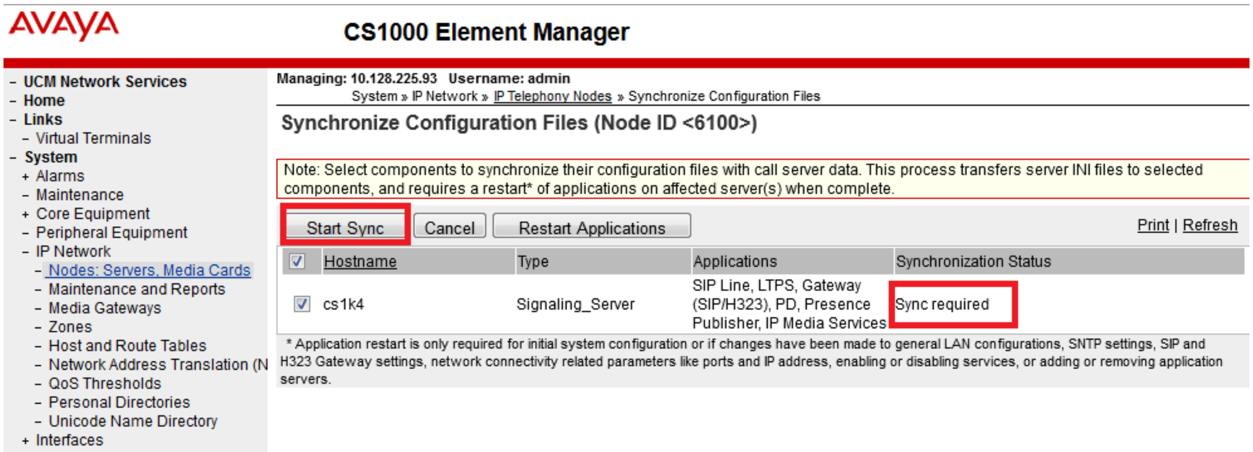
The **Node Details: 6100 – LTPS, Presence Publisher, Gateway Details** page re-appears. Press **Save**.



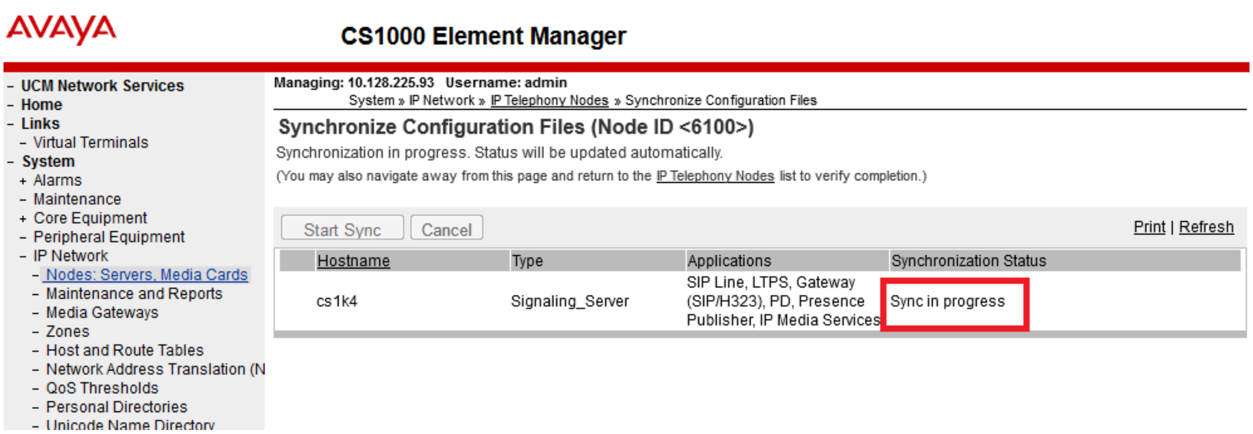
A confirmation message of **Node Saved** appears to indicate that the changes have been saved on the CS 1000 Call Server. Select **Transfer Now**.



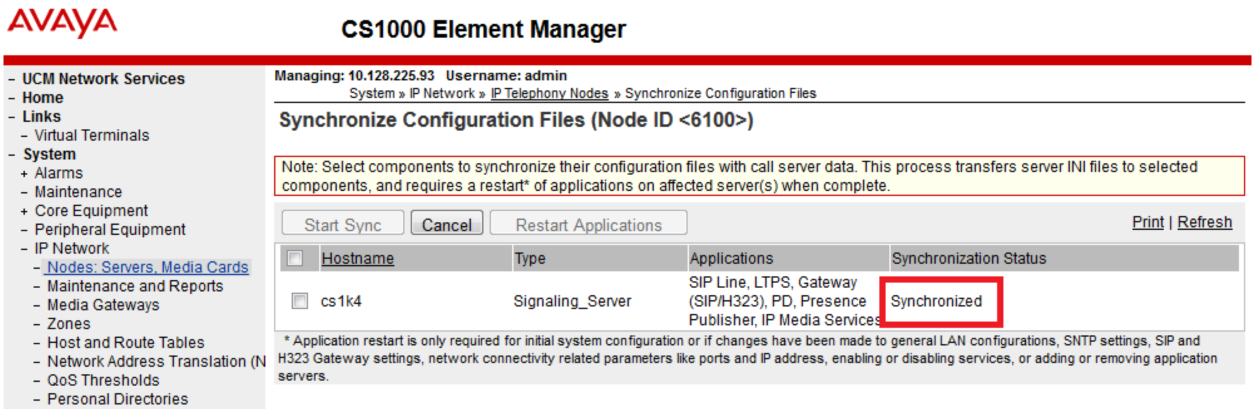
Once the transfer is complete, the **Synchronize Configuration Files (Node ID <6100>)** page appears. Select all the Signaling Servers listed on this page and click on **Start Sync**.



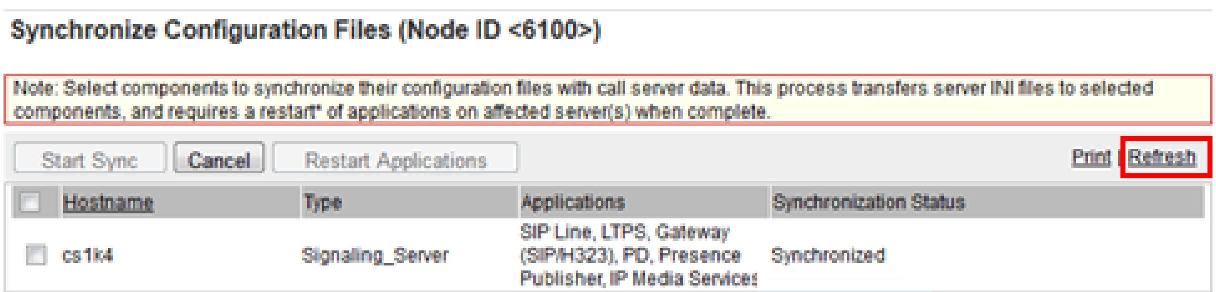
The screen will automatically refresh during the synchronization process. The **Synchronization Status** field will update from **Sync in progress** (as shown for the first Signaling Server) to **Synchronized** (as shown for the second Signaling Server).



After synchronization completes, again select all the Signaling Servers listed on this page and click on **Restart Applications** to use new SIP Gateway settings.



A message will appear stating that **Application restart/reboot has been invoked on selected servers in a synchronized** state to confirm the restart. Click on the **Refresh** button until this message disappears to indicate that the restart has completed.

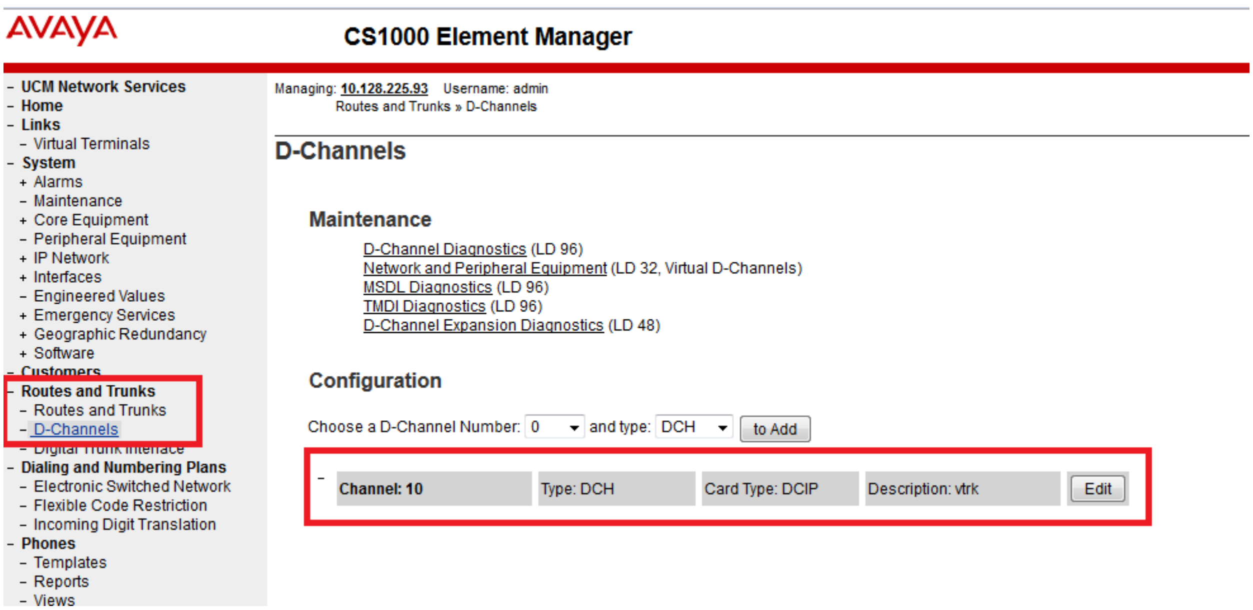


5.5. Confirm Virtual D-Channel, Routes and Trunks

The CS 1000E Call Server communicates with the SIP Gateway (Signaling Server) and Avaya Aura® Session Manager using a virtual D-channel and associated SIP trunk and route. This section describes the steps to verify that this administration has already been completed.

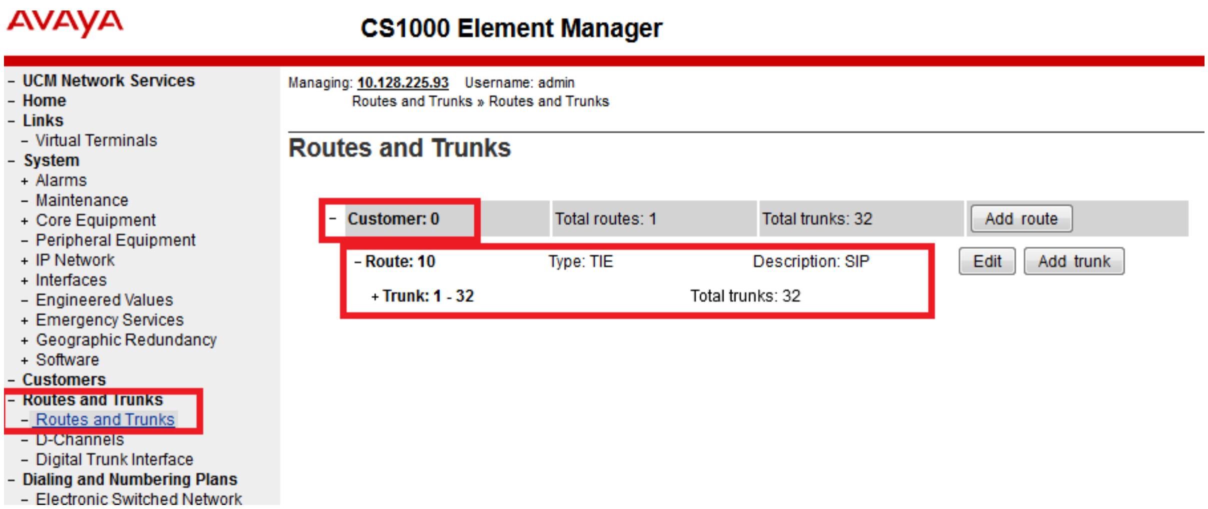
5.5.1. Confirm Virtual D-Channel Configuration

Still in the CS 1000 Element Manager page, expand **Routes and Trunks** list on the left navigation panel and select **D-Channels**. The resulting screen shows all the D-channels configured on the CS 1000 system. In the sample configuration, there is a single D-channel assigned to **Channel: 10** with **Card Type: DCIP**. Specifying **DCIP** as the card type indicates that the D-channel is a virtual D-channel (i.e. D-Channel over IP).

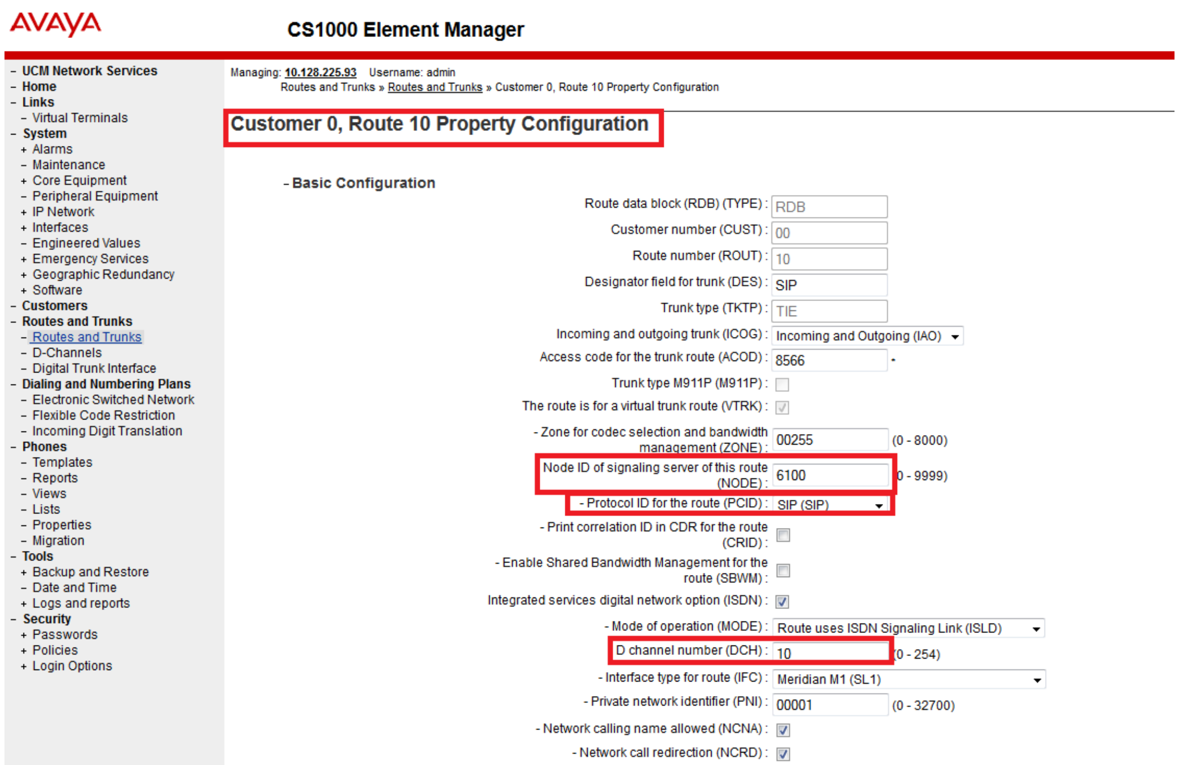


5.5.2. Confirm SIP Route and Trunk Configuration

Still in the CS 1000 Element Manager page, expand the **Routes and Trunks** list on the left navigation panel and select **Routes and Trunks**. The resultant **Routes and Trunks** page appears as shown in the example below. Click on **Customer: 0** to expand its entries. **Route 10** is shown with a description of **VTRK** (for Virtual Trunk). Expand **Route 10** to show that it has been configured with **Total trunks: 32** which indicate that the system is configured to handle 32 simultaneous calls out to Session Manager over SIP. Select **Edit** to verify the configuration of route 10.



The details of the virtual Route 10 defined for the sample configuration is shown below under the heading **Customer 0, Route 1 Property Configuration**. This example confirms that the **Node ID of signaling server of this route (NODE)** is set to **6100** and the **Protocol ID for the route (PCID)** has already been set to **SIP (SIP)**. It also shows that the **D channel number (DCH)** field has been set to match the virtual D-Channel value(**1**) identified above in **section**



5.6. Configure ESN for Route List Index and Digit Manipulation

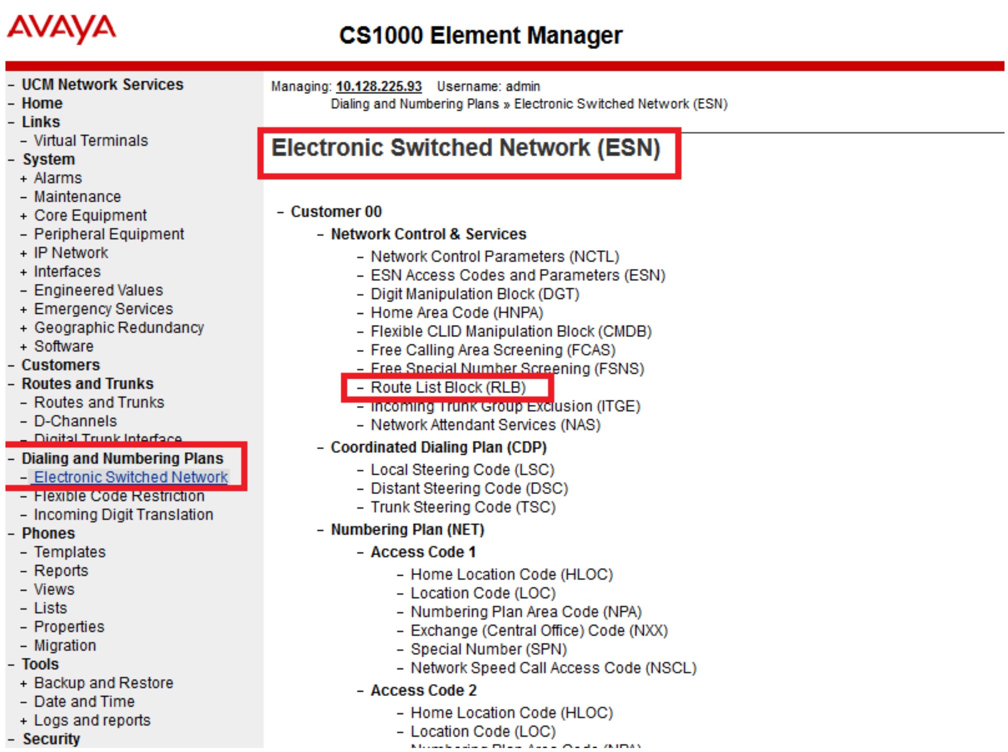
This section provides details of the routing configuration used in the sample configuration to route calls over the SIP Trunk from CS 1000 to Session Manager

Note: The CS 1000 Dialing plan and Aura routing with adaptations will normalize the dial plan allowing Convergence of User endpoints. The routing rules defined in this section are an example and were used in the reference configuration. Other routing policies may be appropriate for different customer networks.

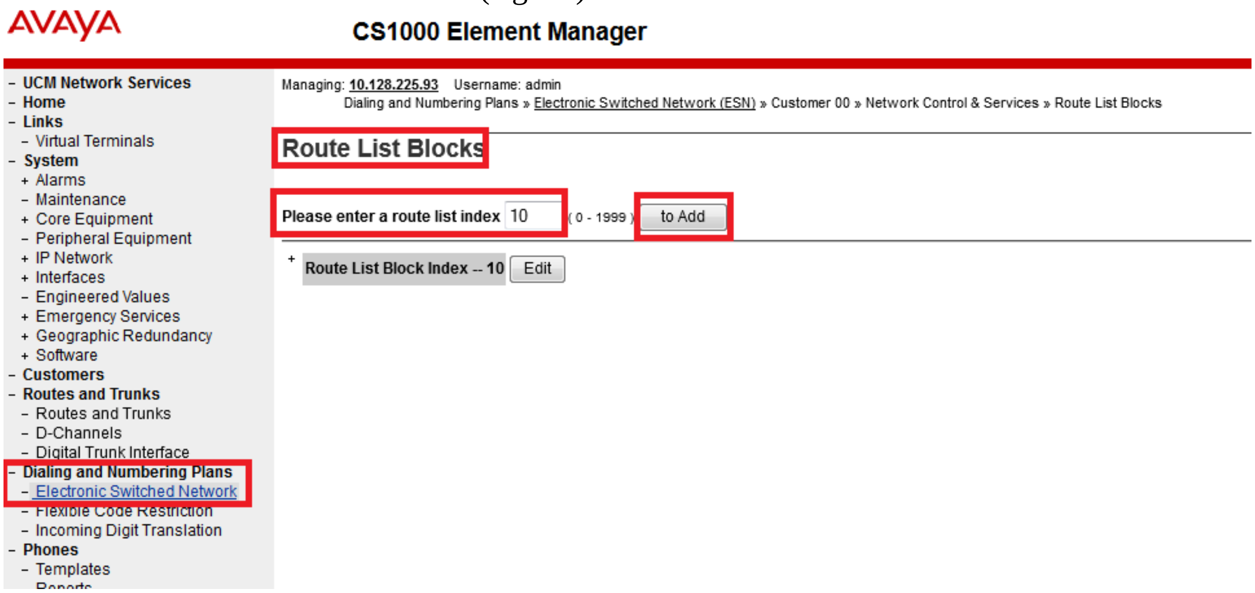
Note: All CS 1000 user before the introduction of Lync Integration have a 5 digit number beginning with 7 (7xxxx). CS 1000 endpoints in the sample configuration used a 5-digit number with all DNs beginning with 7 (e.g. 7xxxx). All Communicator for Lync Integration are provided with a new5-digit number with all DN beginning with 2 (e.g.20xxx). To allow other CS 1000 users to continue to use the 5 digit dial plan all of the Communicator for Lync users will also have a 5-digit virtual extension equal to their existing 5 digits of their endpoint extension (e.g. 7xxxx). All Aura endpoints in the sample configuration used a 7-digit number plan with all DNs Aura DN 447xxxx.

5.6.1. Create Route List Index

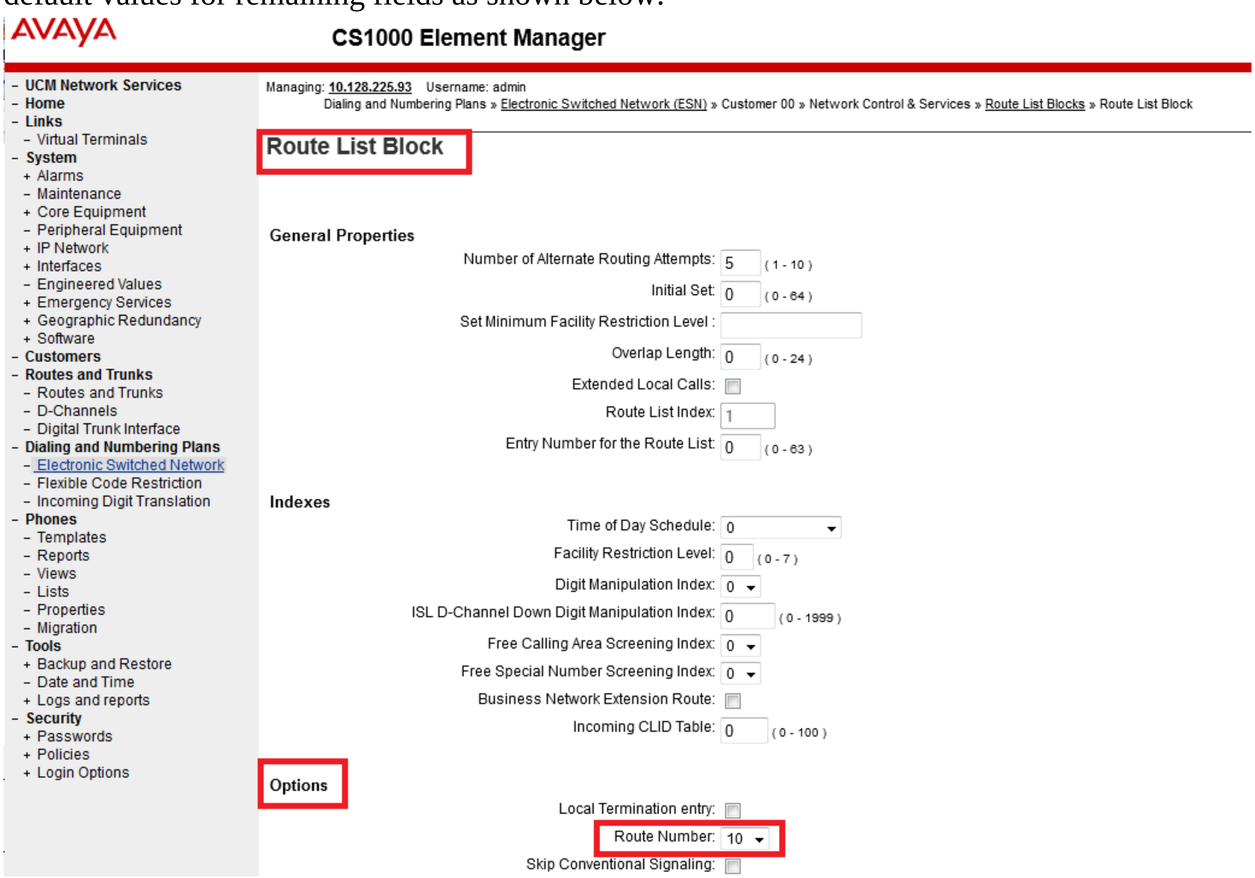
From the **CS 1000 Element Manager** page, expand **Dialing and Numbering Plans** on the left navigational panel and select **Electronic Switched Network**. The **Electronic Switched Network (ESN)** page appears. Click on the link for **Route List Block (RLB)**.



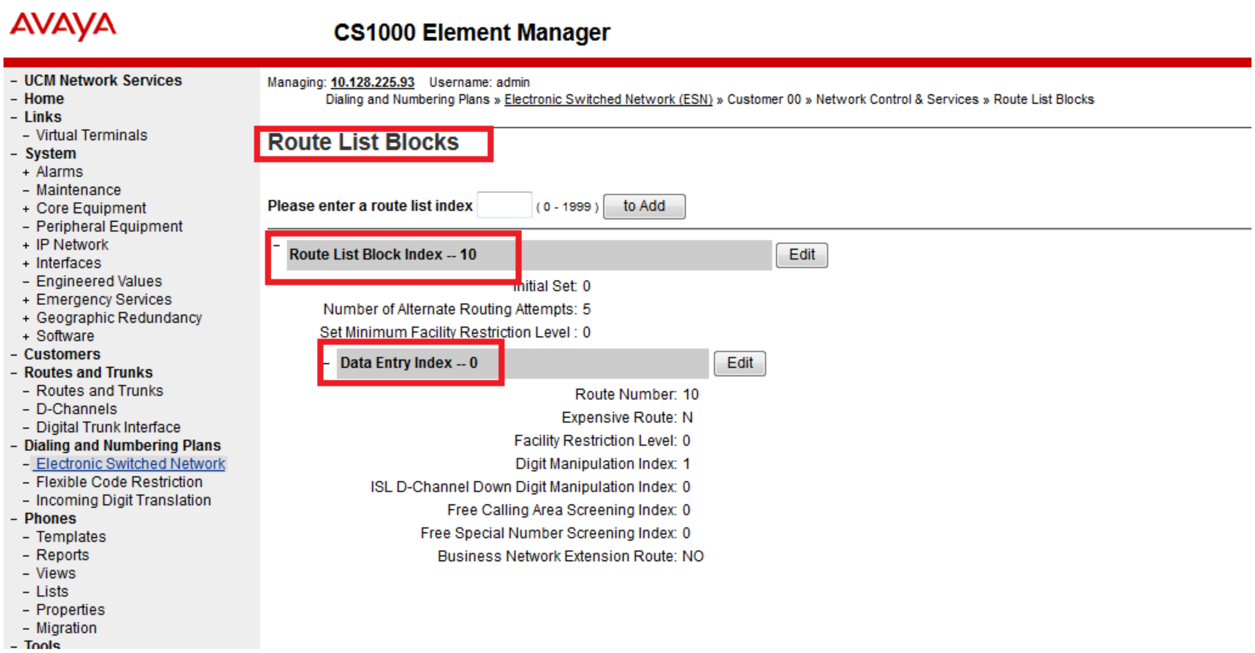
The **Route List Blocks** screen is displayed. Enter an available route list index number in the **Please enter a route list index** field (e.g. **10**) and click **to Add** as shown below.



The **Route List Block** window appears. Under the **Options** section, in the drop-down list, select the **Route Number** of the route identified above in **Section 6.5.2**(i.e. Route Number **10**) and use default values for remaining fields as shown below.



Scroll down to the bottom of this window and click on Submit (not shown) to save the new RLB. The **Route List Blocks** window reappears. Expand the item **Route List Block Index -- 10** and then expand the item **Data Entry Index -- 0** for the new RLB and verify that the entries have been added ok.



5.6.2. Create Distant Steering Code

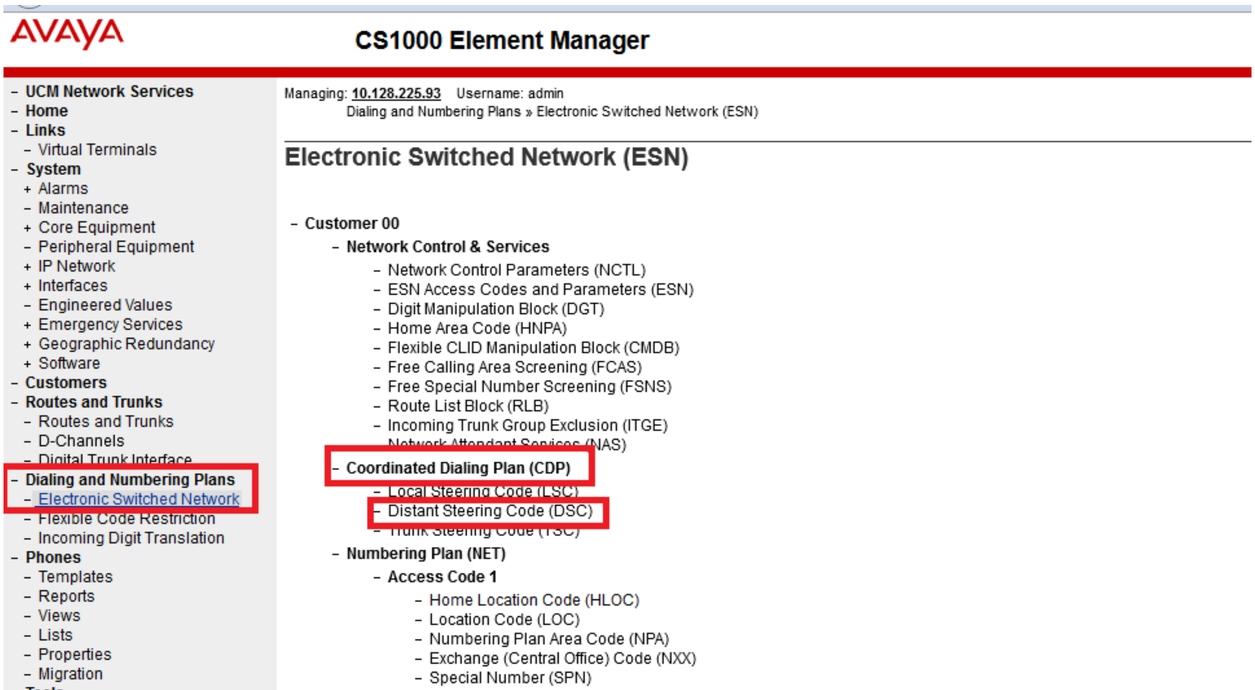
A Distant Steering Code (DSC) digit string will be used as the unique Converged Route Prefix for each CS 1000 user's Personal Call Assistant (PCA) configuration to route voice calls to the Aura Communication Manager via the Session Manager. This DSC will also route Message Waiting Indicator (MWI) messages from CallPilot across the SIP trunk to Communication Manager via the Session Manager. The following table summarizes this as follows:

Number type	Prefix (DSC) / FLEN	RLI	Route
PCA: 447xxxx	447 / 7	10	10

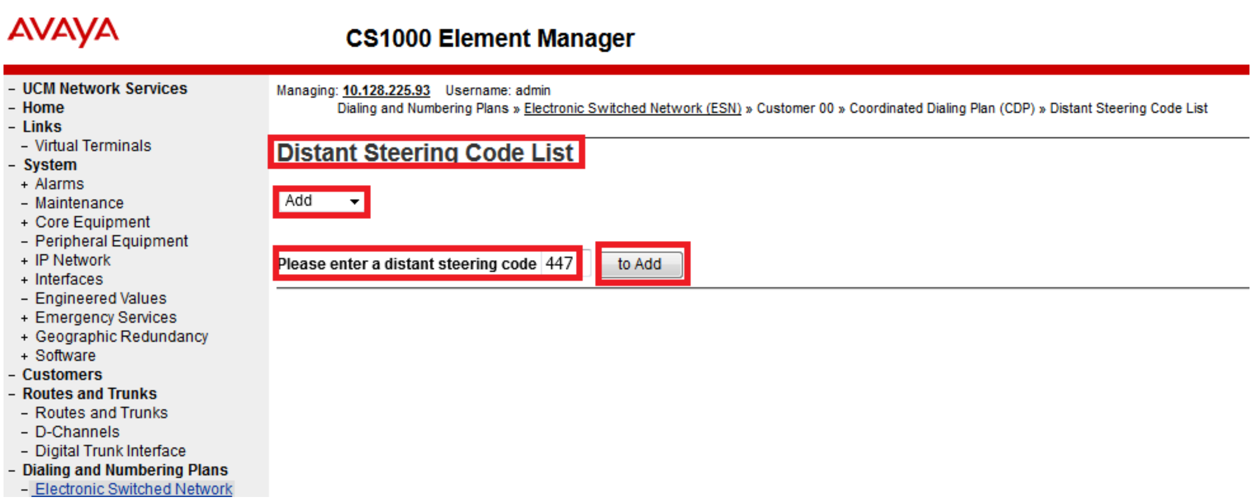
So for example, a call made to CS 1000 virtual DN 70408 will also get sent, using the PCA feature, to CM extension 4470408 . DSC 447, which has a Flexible Length (FLEN) of 7, will send these 7 digits via RLI 1 to route 10 which is the SIP trunk route to Session Manager and on to Communication Manager.
When CallPilot wants to send an MWI to this user upon receiving a new voice mail, it will send the MWI directly to the set on CS 1000 (DN20408) and also to CM 4470408.

From the **CS 1000 Element Manager** page, expand **Dialing and Numbering Plans** on the left navigational panel and select **Electronic Switched Network**. The **Electronic Switched**

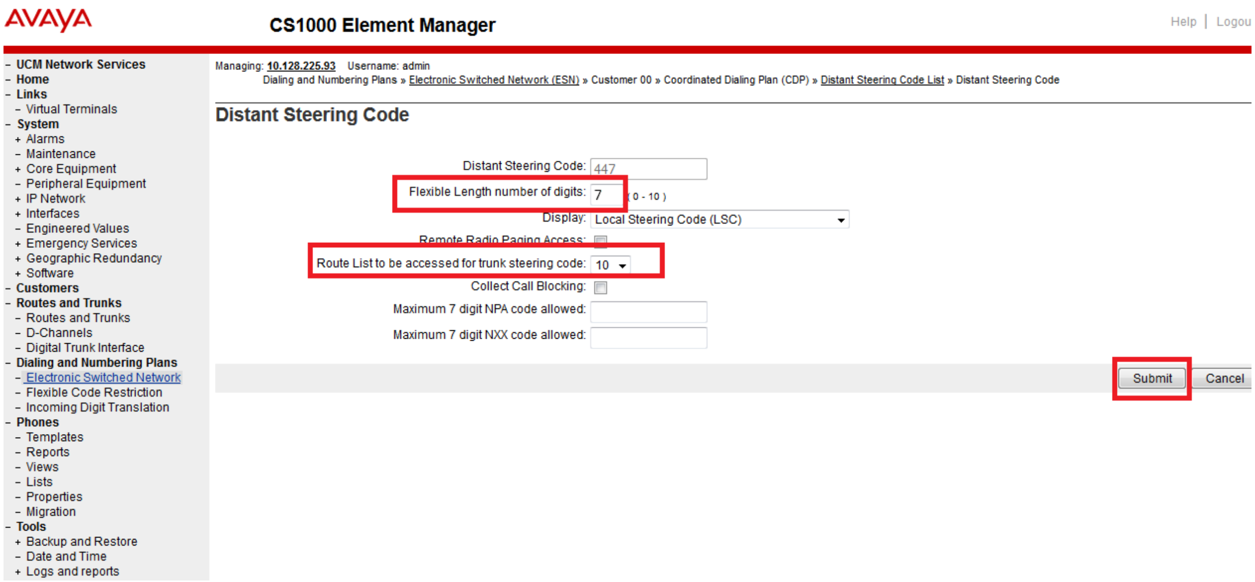
Network (ESN) page appears. Select **Distant Steering Code (DSC)** under the **Coordinated Dialing Plan (CDP)** section.



On the **Distant Steering Code List** page, select **Add** from the drop-down menu. In the box titled **Please enter a distant steering code** enter the dialed prefix for calls to be routed over SIP trunk to Session Manager (e.g. enter **447**). Click **to Add** button to create the new distant steering code.



The **Distant Steering Code** window appears. For **Flexible Length number of digits**, enter **7**. For **Route List to be accessed for trunk steering code**, select the number **10** from the drop-down list. This was the RLI number added in the previous **Section 5.6.1**. Click on **Submit** to save the new Distant Steering Code definition.



The **Distant Steering Code List** page reappears. To view the DSCs just added, enter a DSC into the **Starting Distant Steering Code** box (e.g. **447**) and click on the **View** button. Expand the **Distant Steering Code List – 447**item to show the parameters of the DSC.

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager to receive and route calls over the SIP trunk between CS 1000 and the Avaya Aura

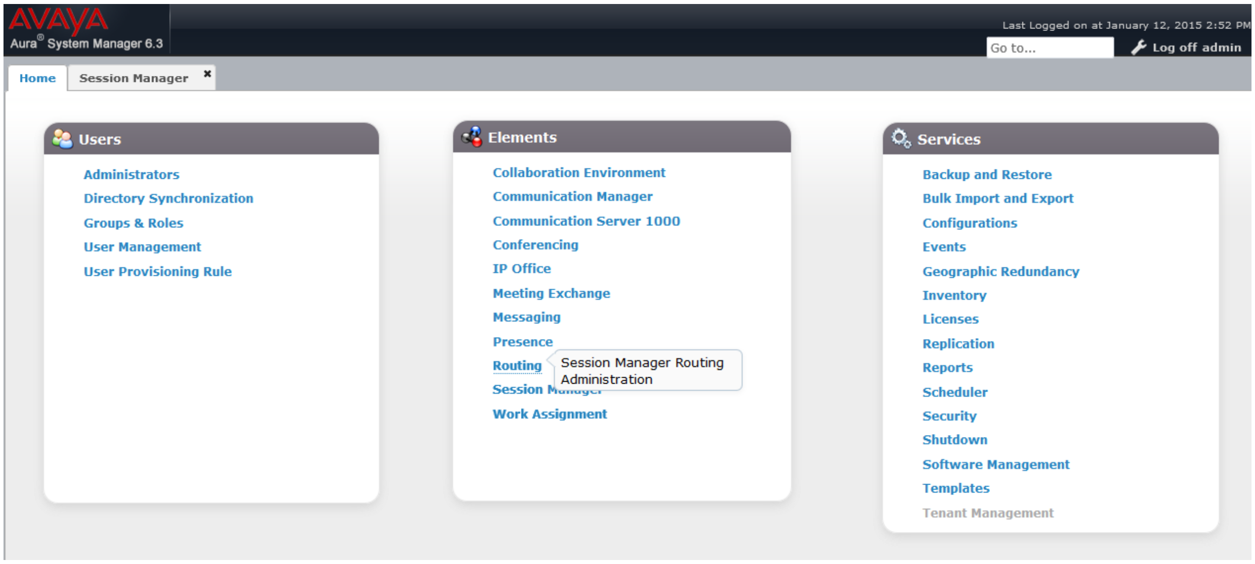
These instructions assume other administration activities have already been completed with the Avaya Midsize Enterprise Template installation such as defining the SIP entity for Session Manager and defining the network connection between System Manager and Session Manager. Upon completion of the Avaya Midsize Enterprise Template installation and configuration, the following Session Manager configuration tasks are required to align with the CS 1000.

Specifically, the following administration activities will be described:

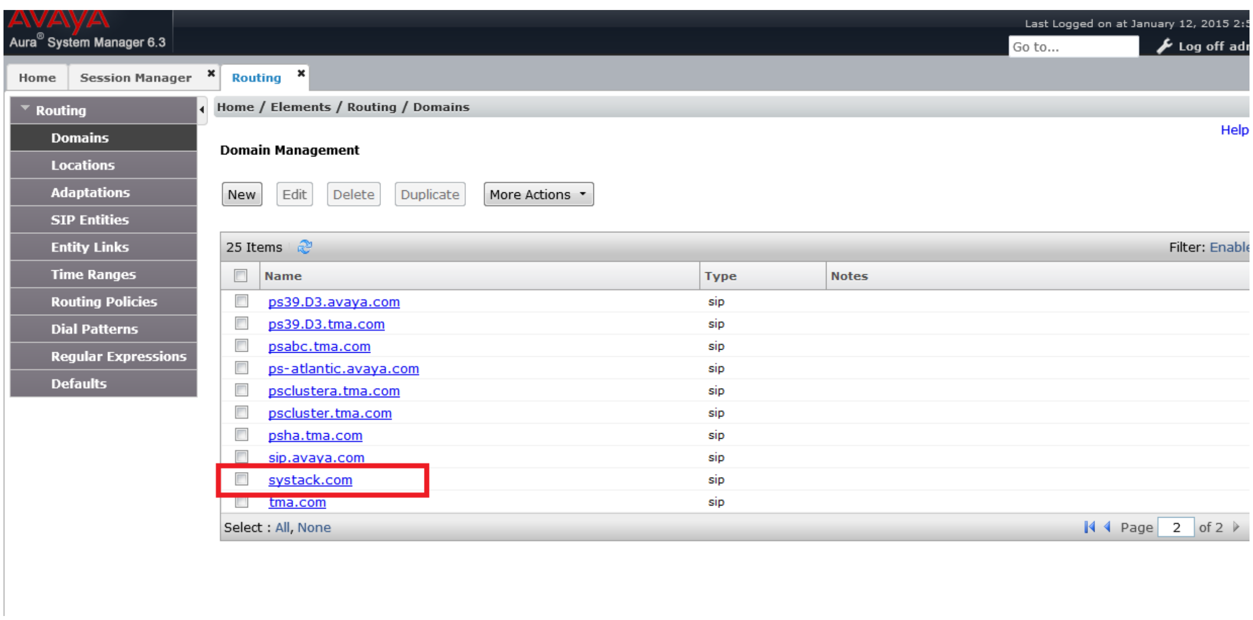
- Verify SIP Domains
- Define a Location for Avaya Communication Server 1000
- Configure Adaptation Module
- Define SIP Entities
    - SIP Entity for CS 1000
    - SIP Entity for Communication Manager
- Define Entity Links
    - Entity Link for Avaya Communication Server 1000
    - Entity Link for Communication Manager
- Define Routing Policy
    - Define the Routing Policy for calls to Avaya Aura® Communication Manager
    - Define the Routing Policy for calls to Avaya Communication Server 1000
- Define Dial Patterns.

6.1. Verify SIP Domains

From the main System Manager page under **Elements**, click on **Routing**.

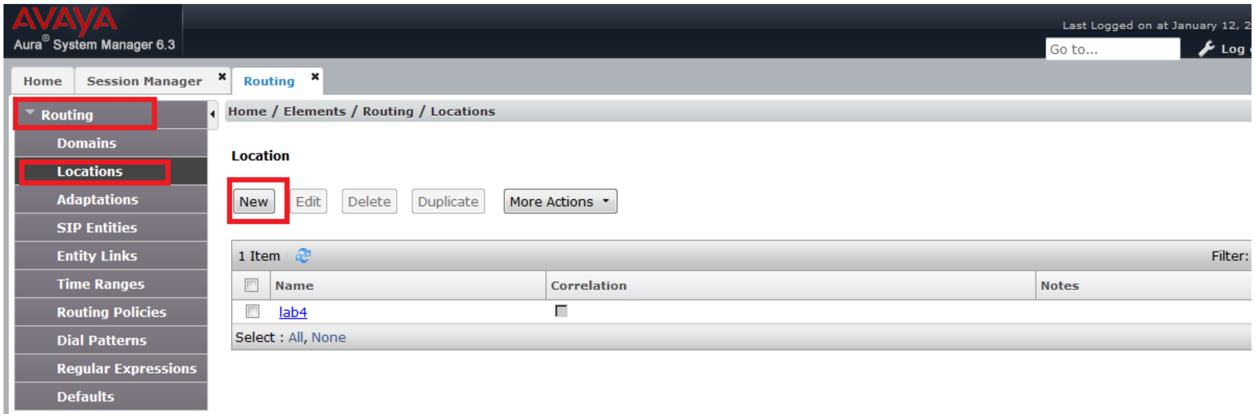


Expand the **Routing** list and select **Domains** from the left navigation menu and verify that the Domain **Name** is the same as the CS 1000 SIP Gateway Domain as shown in **Section 5.4**. In this sample configuration **svstack.com** was used.

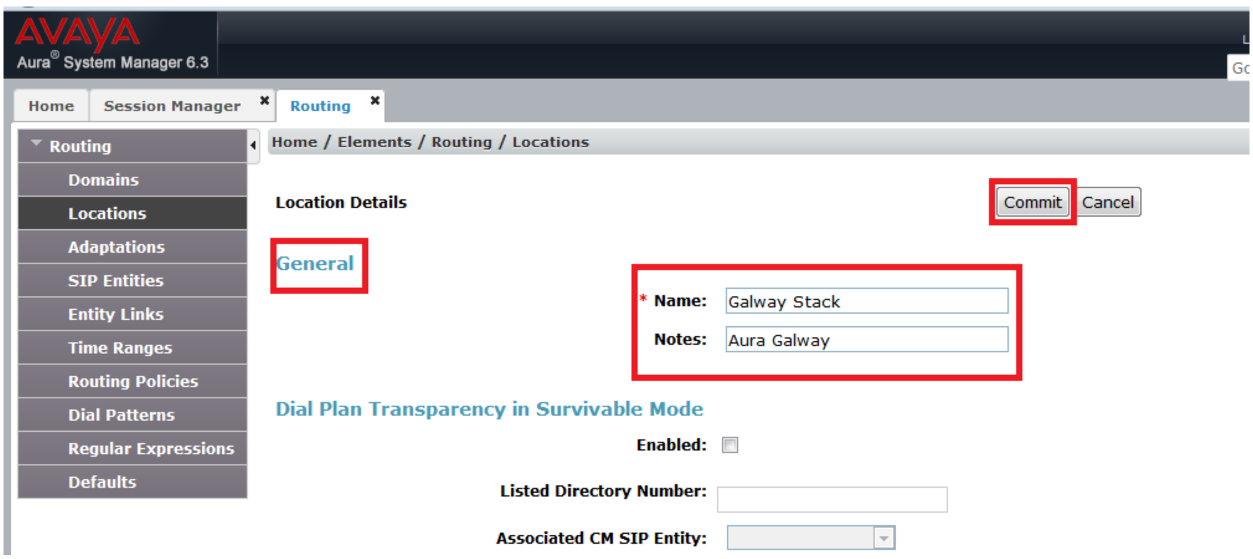


6.2. Define Location for Avaya Communication Server 1000

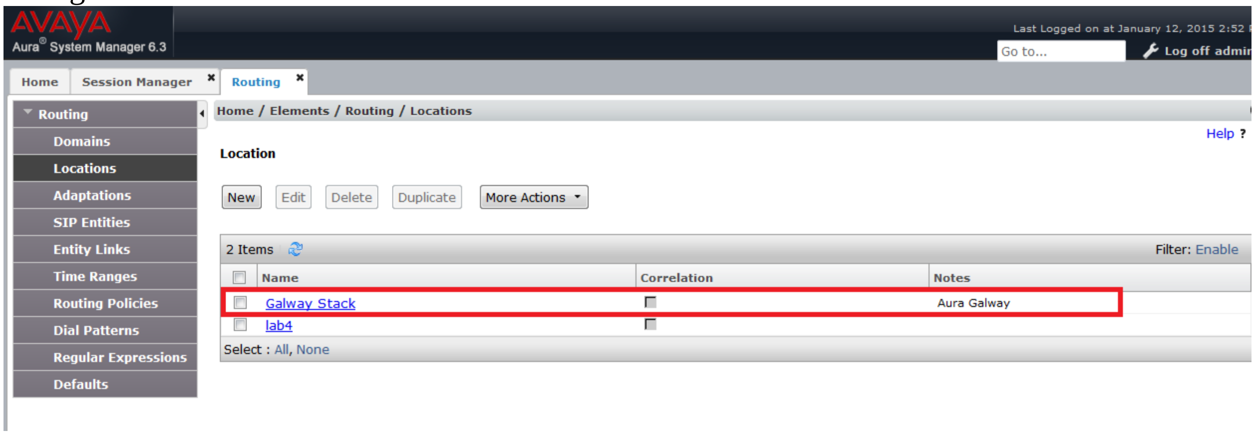
Locations are used to identify the logical and/or physical locations where SIP Entities reside and are also used for the purposes of bandwidth management or location-based routing. Under the **Routing** tab, select **Locations** from the left navigational menu. Click **New**.



In the **General** section, enter a location name in the **Name** box (e.g. **Galway Stack**). Optionally, enter some text into the **Notes** box (e.g. **Aura Galway**). Use the default values for all remaining fields and click **Commit** to save.



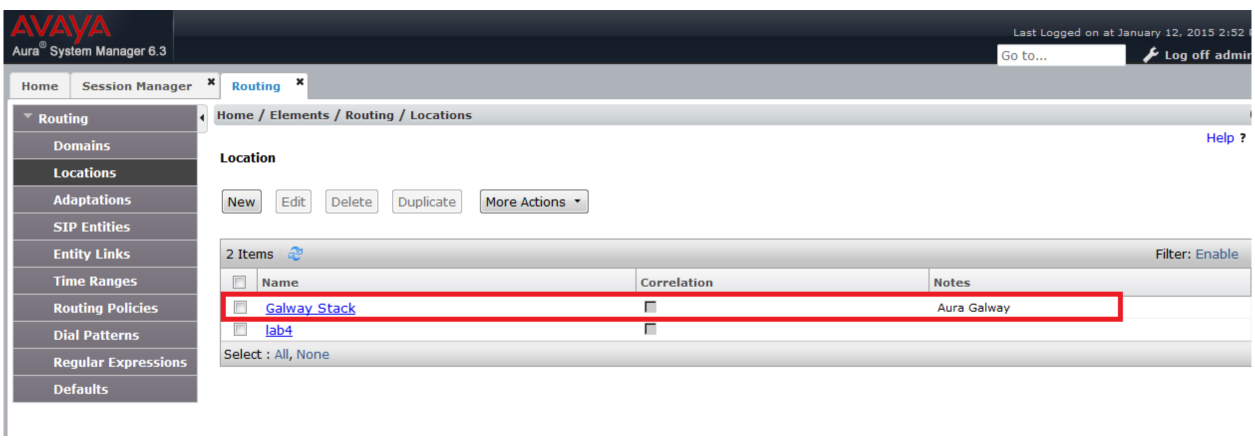
The resultant screen shows the Galway Stack location defined for CS 1000Aura in the sample configuration.



6.3. Configure Adaptation Module

To enable calls between endpoints on CS 1000 and SIP endpoints registered to Session Manager, Session Manager should be configured to use an Adaptation Module designed for CS 1000 to convert SIP headers in messages sent by CS 1000 to the format used by other Avaya products and endpoints. All calls toCS 1000 will have the CS 1000 NARS access code added as they leave Communication Manager (‘1’ is used for this sample configuration). The Session Manager will route all digit strings with a leading ‘1’ to CS 1000. It will be necessary to delete the leading ‘1’ for those calls that terminate within CS 1000 (station calls and Call Pilot).

Under the **Routing** tab, select **Adaptations** from the left navigational menu. Click **New**.



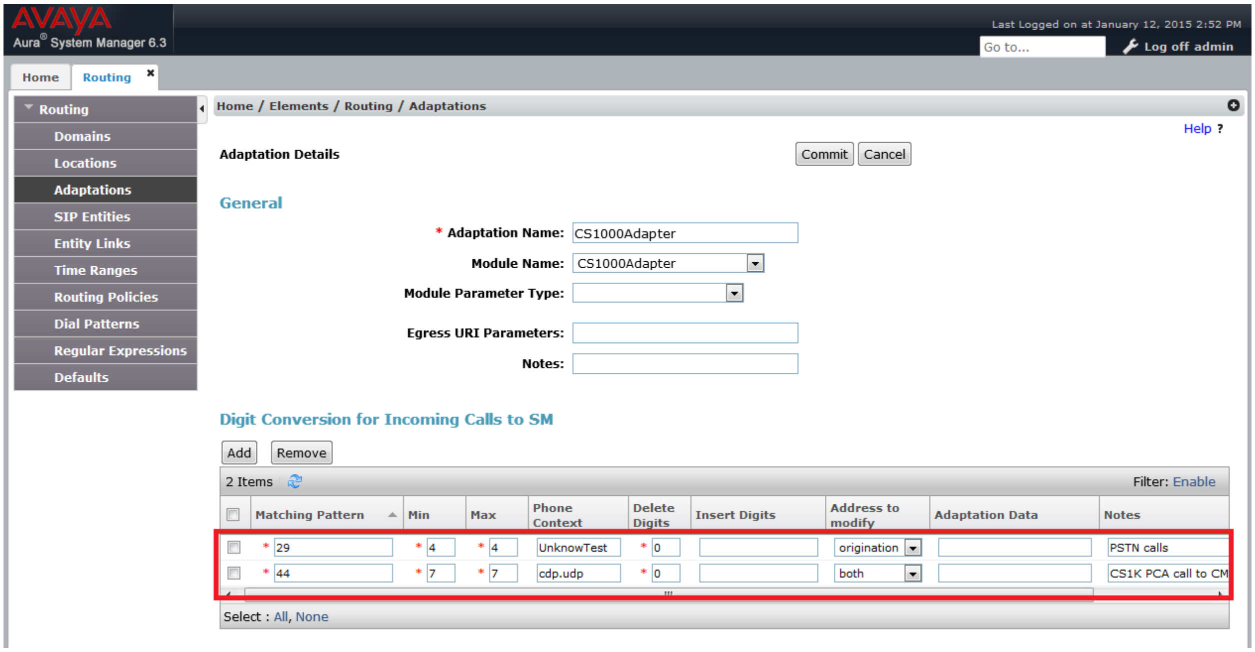
In the **Adaptation Details** page under the **General** section, enter the following values and use default values for remaining fields.

- **Adaptation Name:** Enter an identifier for the Adaptation Module (e.g. **CS 1000**).
- **Module Name:** Select **CS 1000Adapter** from the drop-down menu.
- **Module Parameter:** enter **fromto=true** in this field to ensure that the 'from' SIP header is updated.
- **Notes:** Optionally, enter some descriptive text into this field.

Matching patterns will be used for certain calls to adapt the SIP phone context information for the relevant call. In the sample configuration to simulate a PSTN call, an ISDN trunk was configured between the CS 1000E and a second CS 1000E system where a set with DN 2997 was used as the PSTN endpoint. When calls are made from a Collaborated endpoint to this simulated PSTN, a Private domain name of type Unknown, called "UnknownTest" in the sample configuration, will be sent back from the CS 1000 SIP Gateway to Session Manager which needs to be stripped off using this adaptation (**Note**: this has been configured earlier above as part of the SIP URI Map in **Section 5.4**).So the matching pattern of 29 is used for this purpose. Similarly, when a voice call is made from CS 1000 to a Collaborated endpoint the call is extended over SIP using a PCA and a CDP DSC of 447. Therefore a matching pattern of 44 is used to strip off the phone context of cdp.udp which also was defined earlier in the SIP URI Map in **Section 5.4**. These are summarized in the following table.

Matching Pattern	Min	Max	Phone Context	Delete Digits	Insert Digits	Address to modify	Notes
29	4	4	UnknownTest	0		origination	PSTN Calls
44	7	7	cdp.udp	0		both	CS1k PCA calls to CM

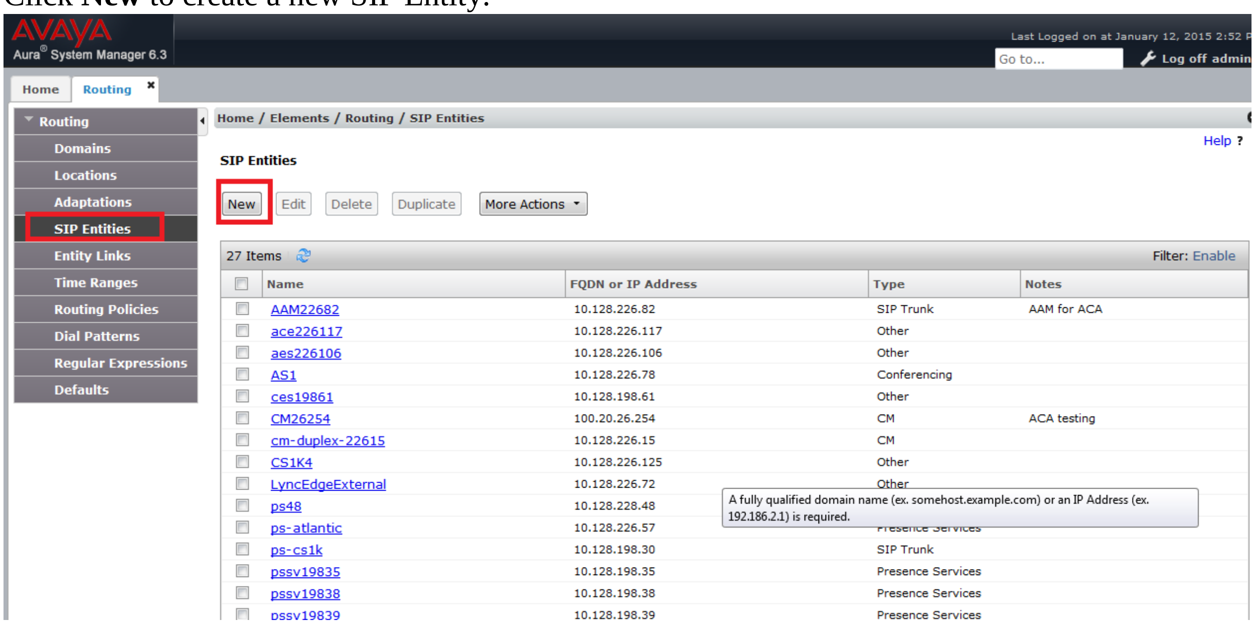
In the **Digit Conversion for Incoming Calls to SM** section, click **Add** and enter these values. It is also recommended to enter optional text in the **Notes** section as a description for each entry.



Click **Commit** to save the changes.

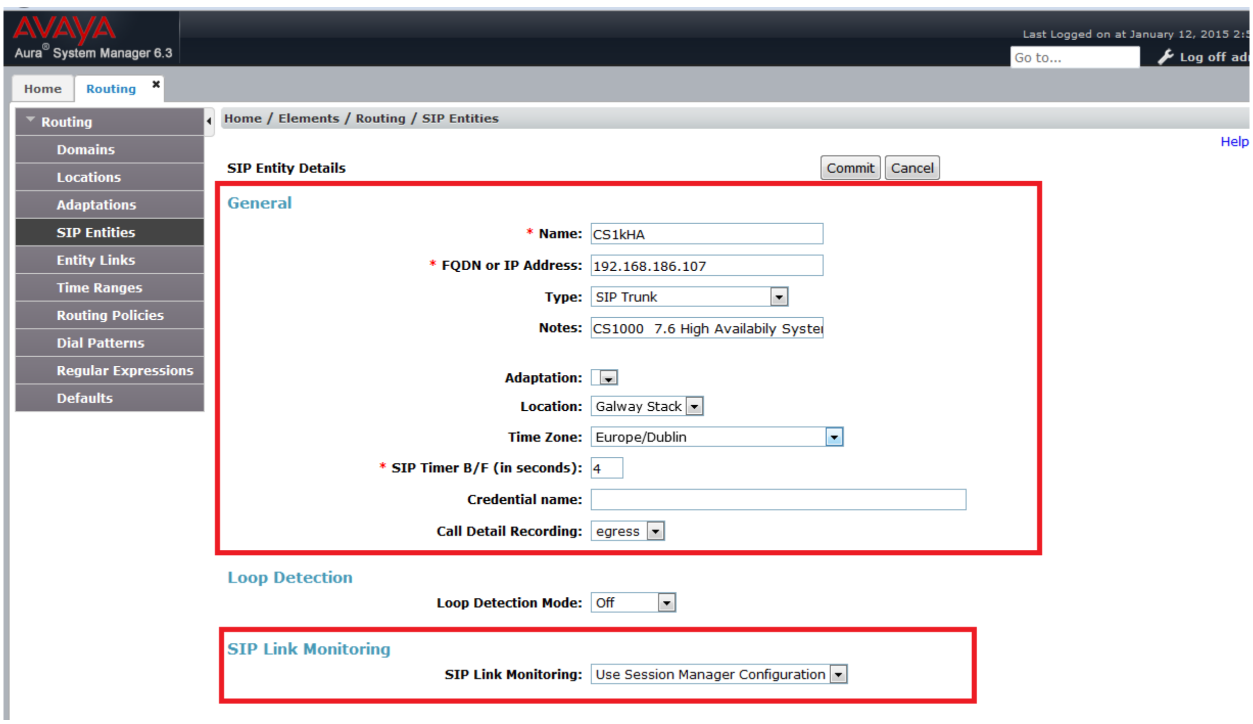
6.4. Define SIP Entities

SIP entities must next be added for CS 1000 and Communication Manager. From the System Manager main page under the **Routing** tab, select **SIP Entities** from the left navigational menu. Click **New** to create a new SIP Entity.



6.4.1. SIP Entity for CS 1000

The **SIP Entity Details** page appears. In the **General** section, enter the following values and use default values for the remaining fields. For **Name**, enter an identifier for the SIP Entity (e.g. **CS1kHA**). For **FQDN or IP Address**, enter the Node IP address of the CS 1000 IP Telephony interface (e.g. **192.168.186.107**).For **Type**, select **SIP Trunk** from the drop-down menu. In the **Notes** box, enter an optional description text (e.g. **CS 1000 7.6 High Availability System**). From the drop-down **Adaptation** list select the **CS 1000** Adaptation Module defined earlier in **Section 5.3**. From the drop-down **Location** list, select the **Galway Stack** Location for CS 1000 as defined earlier in **Section 5.2**. In the **SIP Link Monitoring** section and from the **SIP Link Monitoring** drop-down list**,** select **Use Session Manager Configuration.** Click **Commit** to save the definition of the new SIP Entity.



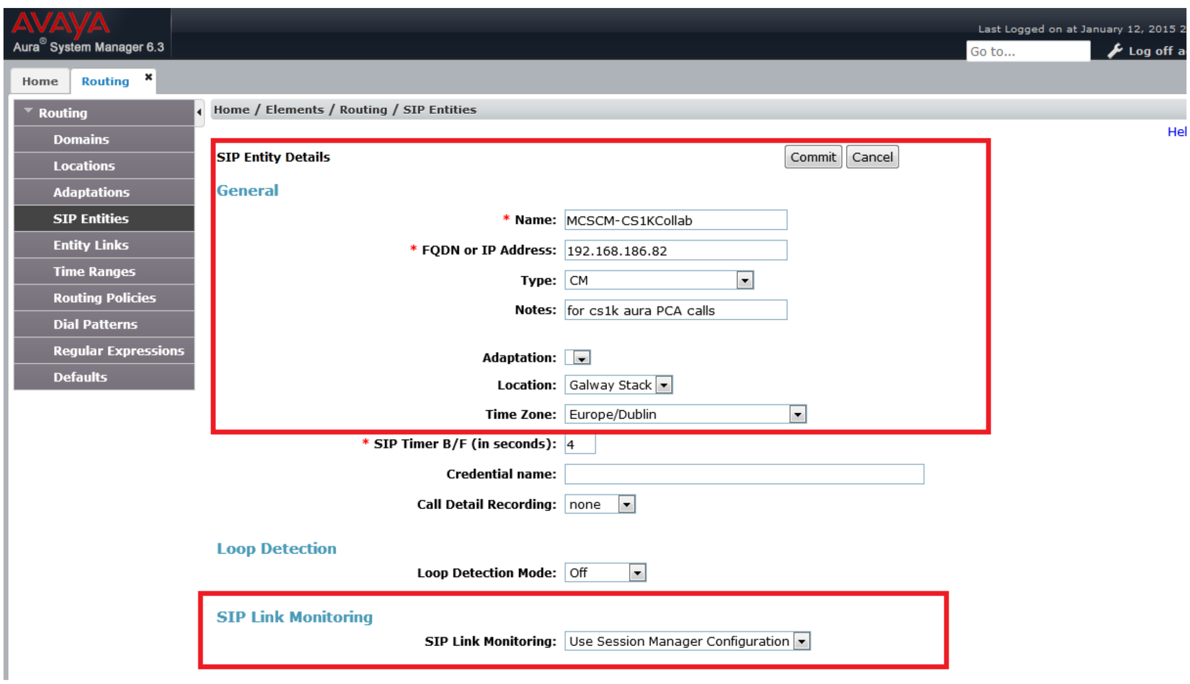
A second SIP Entity is required for the CS 1000 to be associated with the TLS link required to send presence updates from CS 1000 to the Presence server. The name will be different than the first Entity created above and no adaptation will be applied. **Note**: configuration details required for Presence Services are not covered in this Application Note – check references in **Section 10** for more documentation guides for Presence Services.

6.4.2. SIP Entity for Communication Manager

The SIP Entity built during the installation of the Midsize Enterprise (ME) server is required for Collaboration clients to access Communication Manager for features and is dedicated to IMS (IP Multimedia Subsystem) functionality. A second SIP Entity is built for Communication Manager to handle Enterprise traffic (calls to/from CS 1000 and calls to PSTN via CS 1000). This Entity will have the same IP address as the SIP Entity built during the ME server installation but it will use a different TCP port (5062) and therefore use a different

Communication Manager SIP trunk than the SIP Entity and Entity Link built during the ME server install. This is done to ensure that Enterprise traffic can be handled separately from Collaboration client feature verification traffic and so that adaptations applied to Enterprise calling do not interfere with the IMS process. For the sample configuration, the second SIP Entity is named **MESCM-CS1kCollab**.

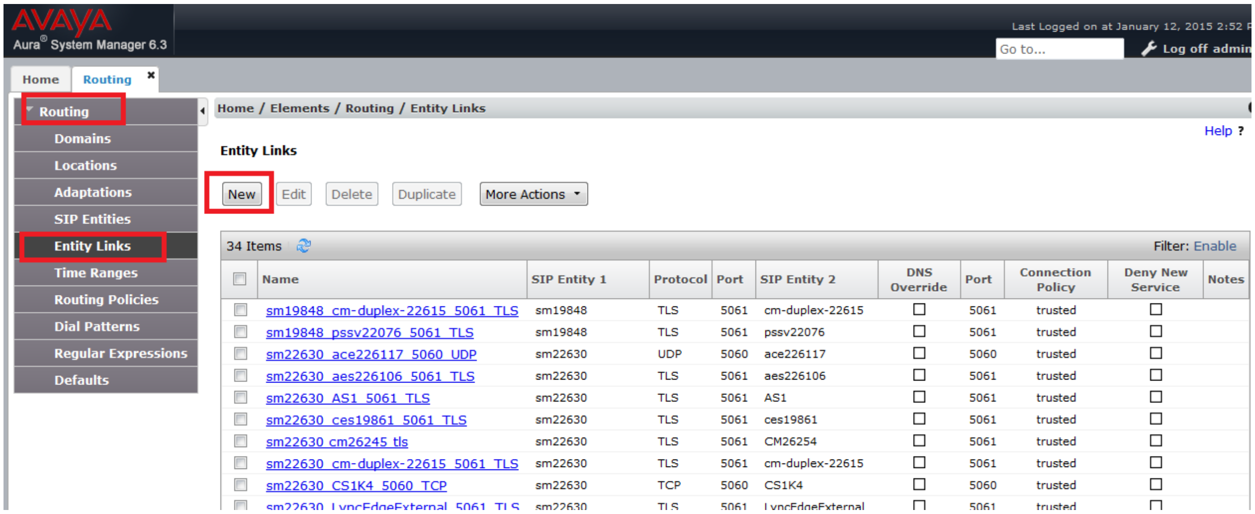
From System Manager main page under the Routing tab, select SIP Entities from the left navigational menu and click New to create a new SIP Entity (as shown above in **Section 5.4**).The **SIP Entity Details** page appears. In the **General** section, enter the following values and use default values for the remaining fields. For **Name**, enter an identifier for the SIP Entity (e.g. **MESCM-CS1kCollab**). For **FQDN or IP Address**, enter the IP address of the Communication Manager (e.g. **192.168.186.82**). For **Type**, select **CM** from the drop-down menu. In the **Notes** box, enter an optional description text (e.g. **For CS1k Aura PCA Calls**). From the drop-down **Location** list, select the **Galway Stack** Location as defined earlier in **Section 5.2**. In the **SIP Link Monitoring** section and from the **SIP Link Monitoring** drop-down list, select **Use Session Manager Configuration.** Click **Commit** to save the definition of the new SIP Entity.



6.5. Define Entity Links

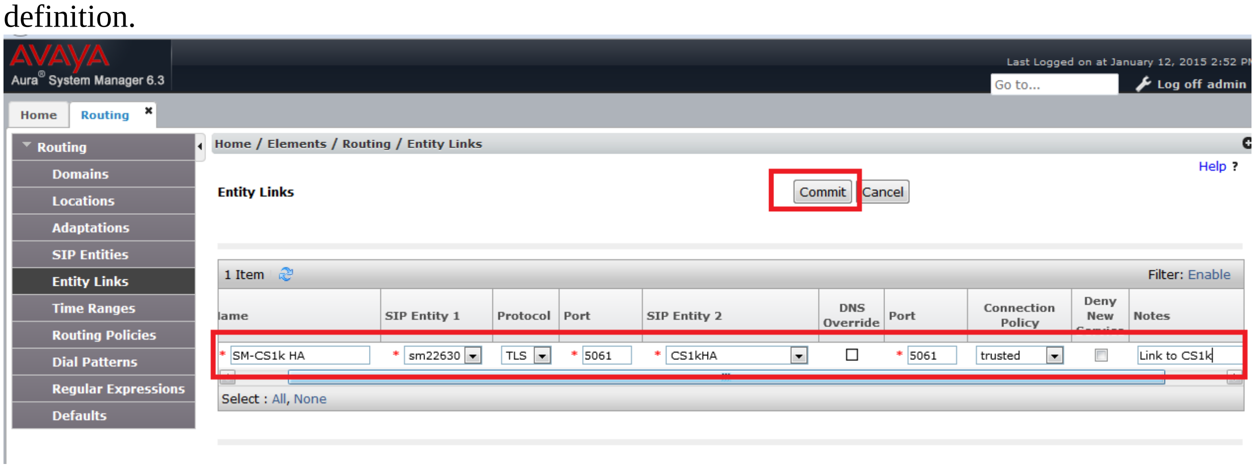
Any connections to Session Manager are described by an Entity Link. In the sample configuration there are two Entity Links between Session Manager and CS 1000, one for MWI/Voice traffic and one for Presence Services. Likewise, two Entity links are required between Session Manager and Communication Manager, one for Collaboration client IMS services (created during the ME server installation) and one for communication with the CS 1000.

From the System Manager main page under the **Routing** tab, select **Entity Links** from the left navigational menu. Click **New** to create a new Entity Link.



6.5.1. Entity Link for Avaya Communication Server 1000

The SIP trunk between Session Manager and CS 1000 is described by an Entity link. In the **Entity Links** page, for **Name**, enter an identifier for the link to each telephony system (e.g. **MESSM-CS1kHA**). From the **SIP Entity 1** drop-down list, select the SIP Entity defined for Session Manager (e.g. **MESSM**). From the **SIP Entity 2** drop-down list, select the SIP Entity defined for CS 1000 for voice calls in **Section 5.4** (e.g. **CS1kHA**). From the **Protocol** drop-down list, after selecting both SIP Entities, select **TLS** as the required protocol. Verify that the default listen values in the **Port** fields for both SIP entities have been automatically set to **5061** for TLS. Also verify that the default Connection Policy is set to **Trusted**. Optionally enter a brief description in the **Notes** field (e.g. **Link to CS1k**).Click **Commit** to save the Entity Link definition.

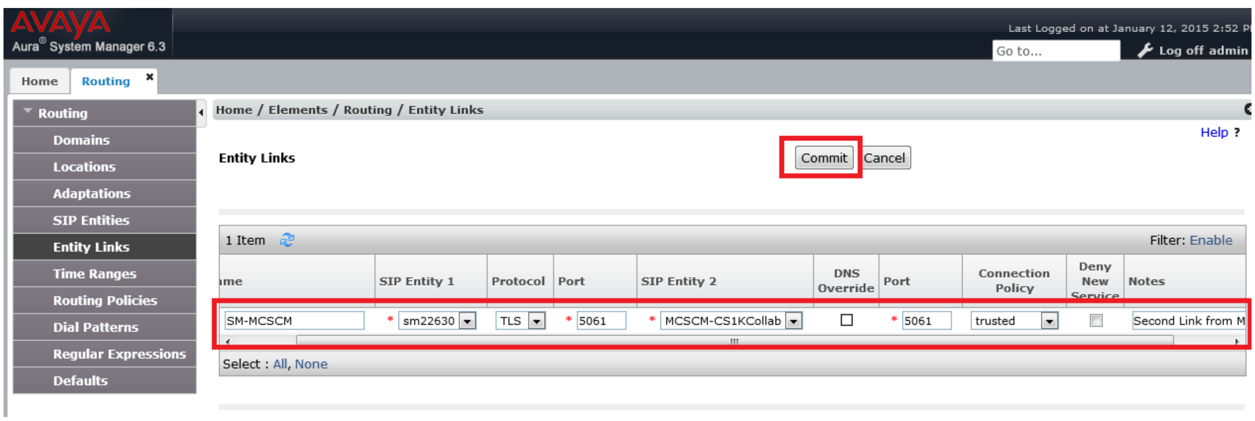


6.5.2. Entity Link for Communication Manager

The Entity Link built between Session Manager and Communication Manager during the installation of the ME server is required for the Collaboration clients to access the Communication Manager for telephony features. A second Entity Link is required between Session Manager and Communication Manager to handle Enterprise traffic (calls to/from

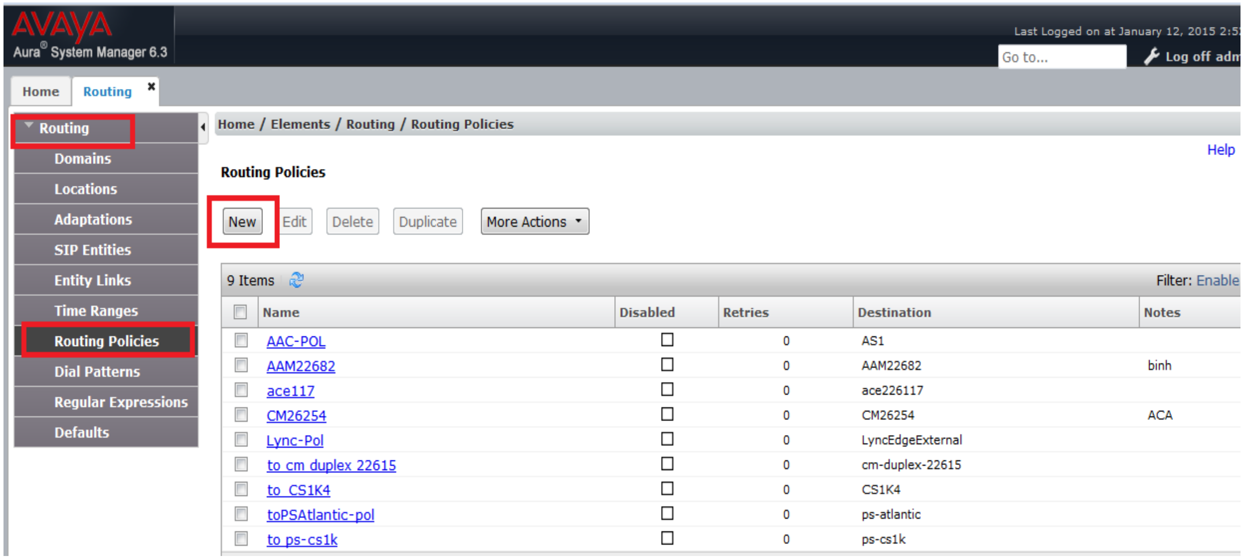
CS 1000 and calls to PSTN via CS 1000). This link will use a different TLS port (e.g. 5062) than that used for the Entity Link built during the ME server installation. Using a different port requires a second SIP trunk to be built in Communication Manager and ensures that any inbound or outbound digit manipulation does not affect the IMS traffic required by the Collaboration clients to function properly.

From the System Manager main page under the Routing tab, select Entity Links from the left navigational menu and click New to create a new Entity Link (as shown above in **Section 5.5**).In the **Entity Links** page, for **Name**, enter an identifier for the link to each telephony system (e.g. **SM-CM-5062**). From the **SIP Entity 1** drop-down list, select the SIP Entity defined for Session Manager (e.g. **MESSM**). From the **SIP Entity 2** drop-down list, select the SIP Entity defined for CM for voice calls in **Section 5.4** (e.g. **MESCM-CS1kCollab**). From the **Protocol** drop-down list, after selecting both SIP Entities, select **TLS** as the required protocol. Enter non-default listen values in the **Port** fields for both SIP entities (e.g. **5062**) for TLS. Also verify that the default Connection Policy is set to **Trusted**. Optionally enter a brief description in the **Notes** field (e.g. **Second Link from MESM to CM**).Click **Commit** to save **Entity Link** definition.

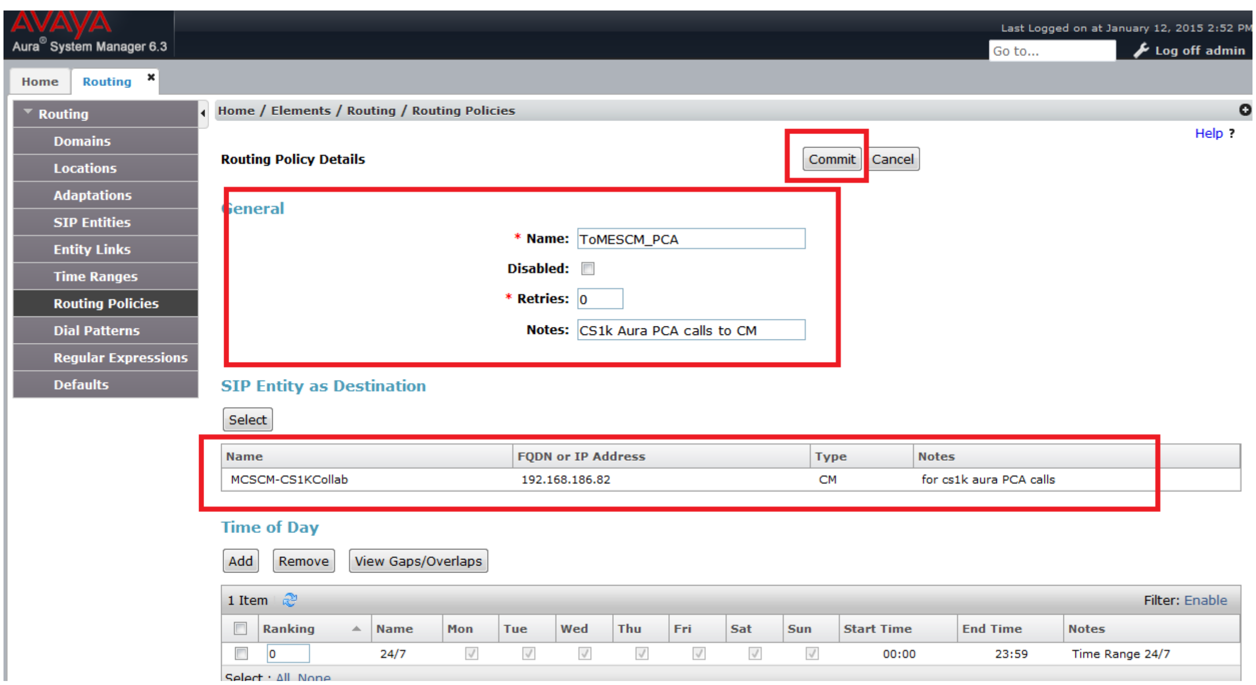


6.5.3. Define Routing Policy

Routing policies describe the conditions under which calls will be routed to CS 1000 from SIP endpoints registered to Session Manager. Routing Policies will also be used to send calls and Callpilot MWI notification messages from the CS 1000 to Communication Manager. From the System Manager main page under the **Routing** tab, select **Routing Policies** from the left navigational menu. Click **New** to create a new Routing Policy.

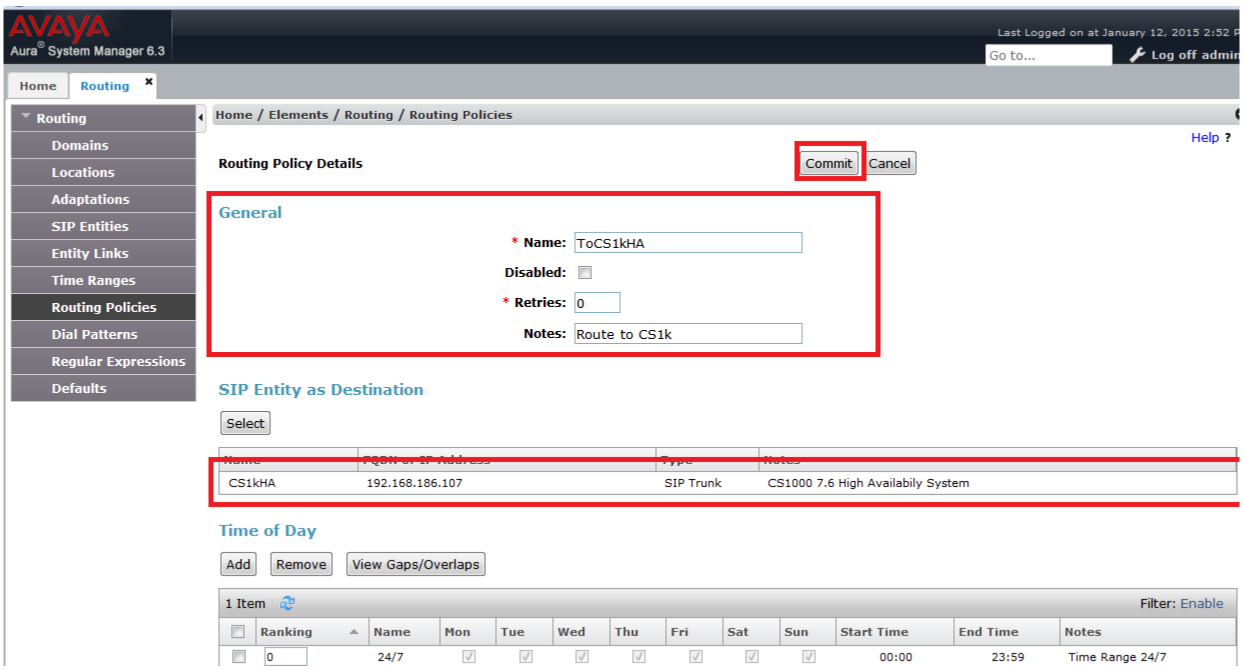


Define the Routing Policy for calls to Avaya Aura® Communication Manager
In the **General** section, enter the following values. For **Name**, enter an identifier to define the routing policy (e.g. **ToMESCM_PCA**). Leave the **Disabled** box unchecked. Optionally, enter some descriptive text in the **Notes** box (e.g. **CS1k Aura PCA calls to CM**). In the **SIP Entity as Destination** section, click **Select.** The SIP Entity List page opens (not shown).Select the SIP Entity associated with Communication Manager as defined above in **Section 5.4.2** (e.g. **MESCM-CS1kCollab**) and click Select. The selected SIP Entity displays on the **Routing Policy Details** page. Use default values for remaining fields. Click **Commit** to save Routing Policy definition.



6.5.4. Define the Routing Policy for calls to Avaya Communication Server 1000

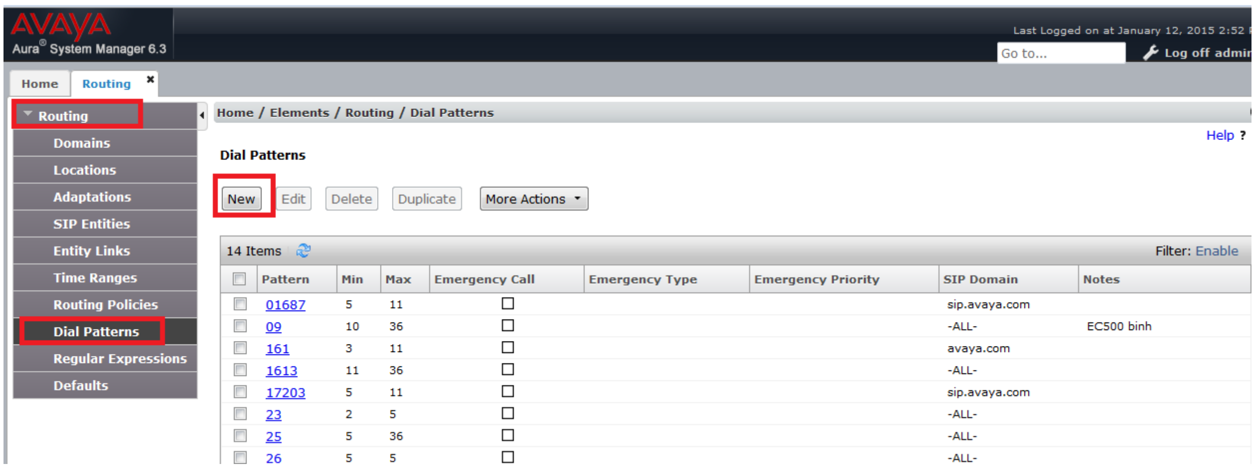
Following the same procedure as above in **Section 5.6.1**, a Routing Policy is built toCS 1000. In the **General** section, enter the following values. For **Name**, enter an identifier to define the routing policy (e.g. **ToCS1kHA**). Leave the **Disabled** box unchecked. Optionally, enter some descriptive text in the **Notes** box (e.g. **Route to CS1k**). In the **SIP Entity as Destination** section, click **Select.** The SIP Entity List page opens (not shown). Select the SIP Entity associated with CS 1000 as defined above in **Section 5.4.1** (e.g. **CS1kHA**) and click Select. The selected SIP Entity displays on the **Routing Policy Details** page. Use default values for remaining fields. Click **Commit** to save Routing Policy definition.



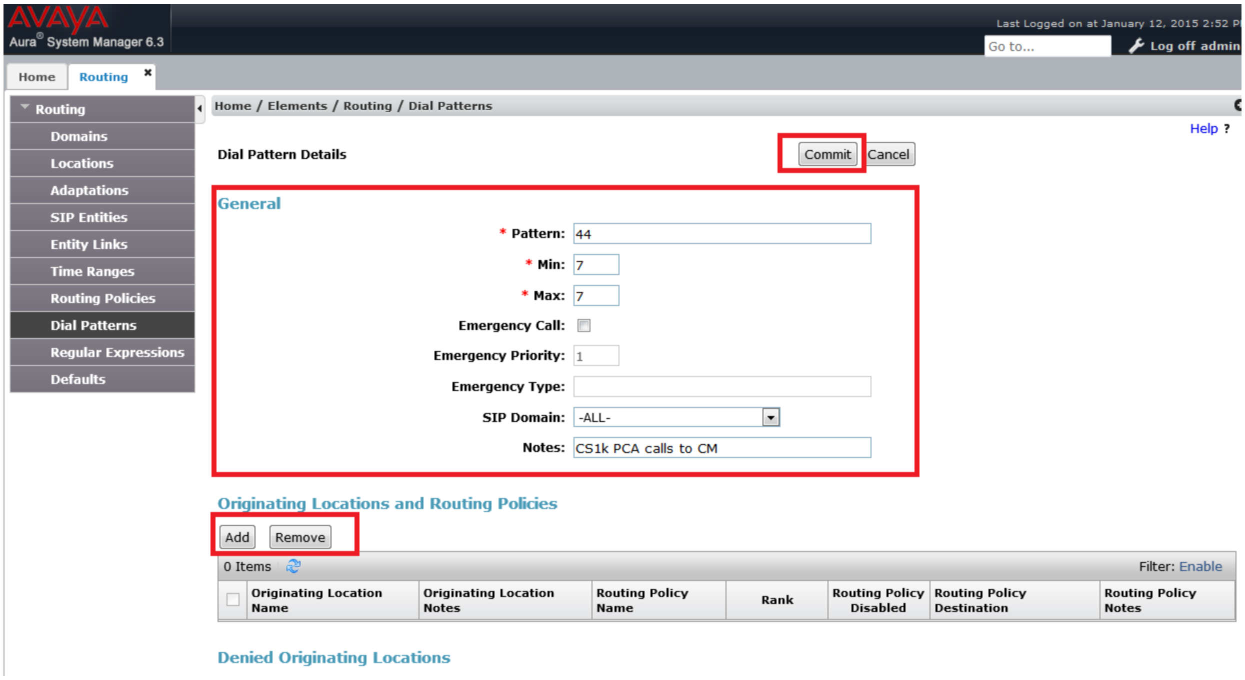
6.6. Define Dial Pattern

Dial patterns are used to route calls to appropriate SIP Entities. In the sample configuration, voice calls from CS 1000 to Communication Manager are extended by PCA with a prefix of 44 in front of the 5 digit DN. So a dial pattern of 44 is used to route these calls to Communication Manager. Calls from the Aura stations to CS 1000 are routed first to Communication Manager per the user profile application sequencing configuration. As CS 1000 extension has a prefix of 7 or 2, therefore dial patterns of 7 and 2 are used to route calls to CS 1000. Other Dial Patterns required for PSTN bound traffic and emergency dialing can be configured using the same steps described below.

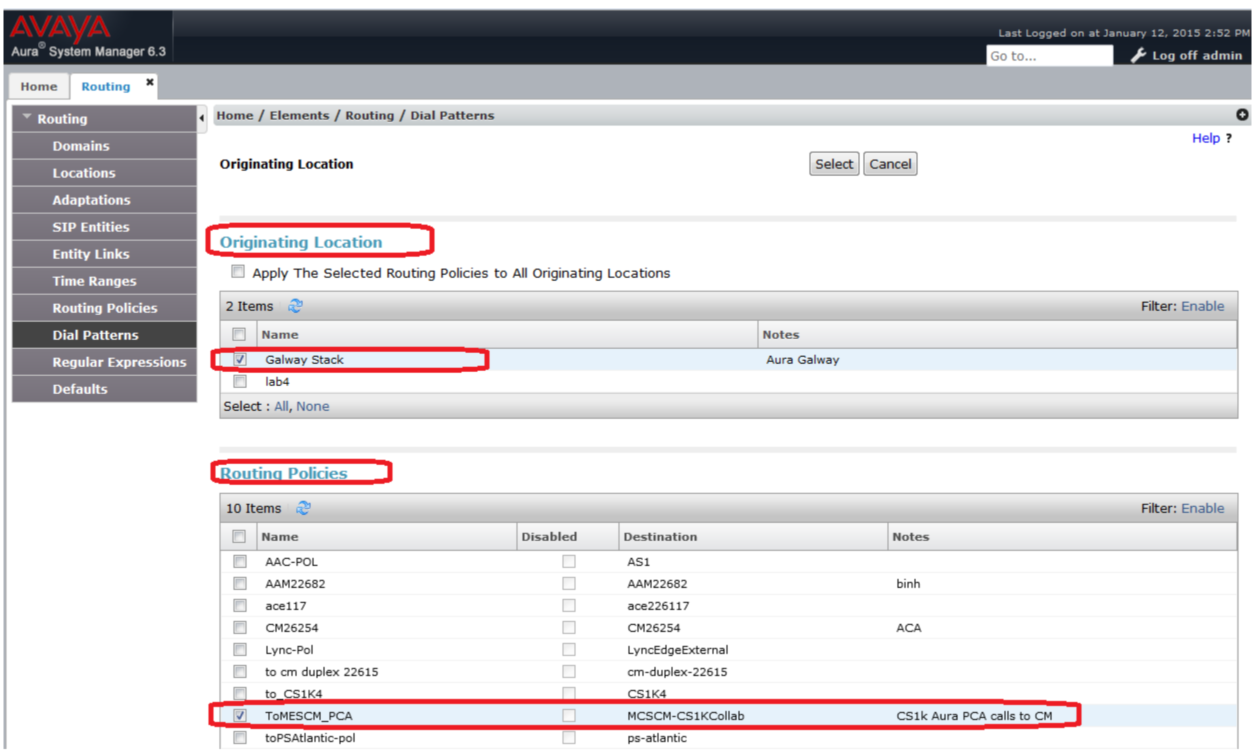
From the System Manager main page under the **Routing** tab, select **Dial Patterns** from the left navigational menu. Click **New** to create a new Dial Pattern.



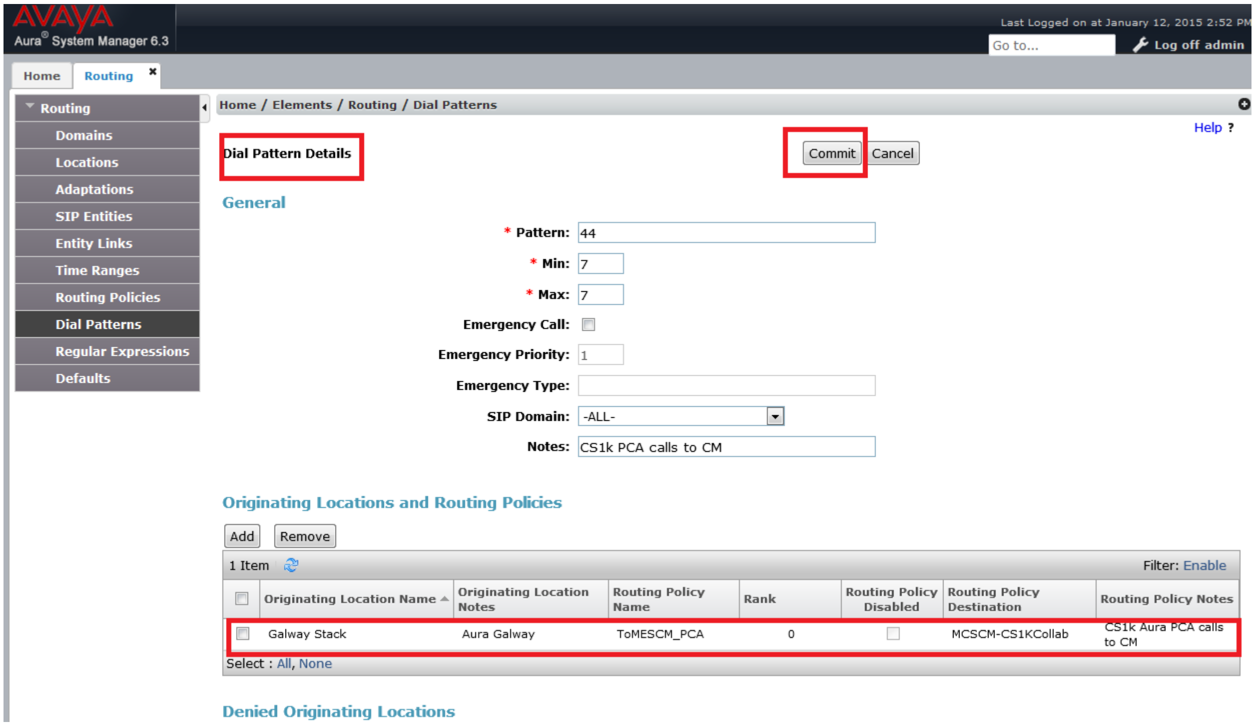
Configure a dial pattern of 44 to be used to route CS 1000 voice calls to Communication Manager as follows. In the **General** section, enter the dial **Pattern** for calls to Communication Manager (e.g. **44**). Because all voice calls to Communication Manager will be 7 digits in length, enter **7** in the **Min** and **Max** fields. In the drop-down **SIP Domain** list, select the SIP domain or select **All** (where Session Manager can accept incoming calls from all SIP domains). Optionally, enter a brief description in the Notes field (e.g. **CS1k PCA calls to CM**). In the **Originating Locations and Routing Policies** section, click **Add**.



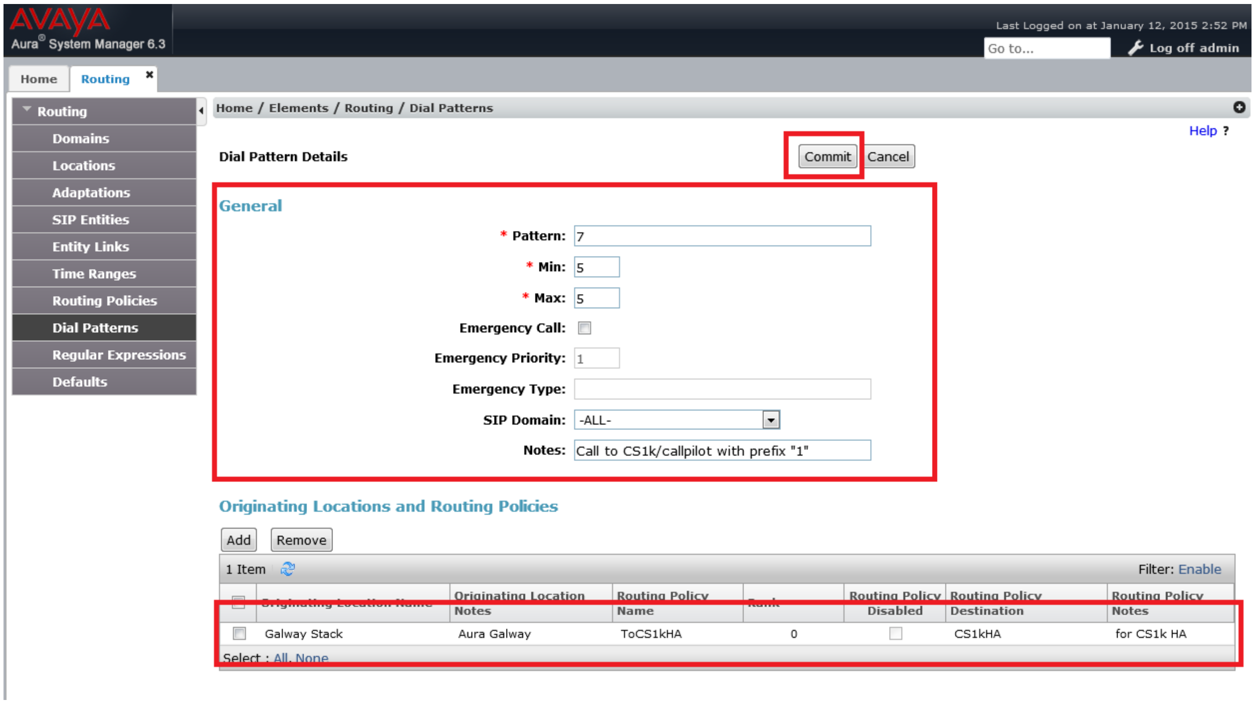
The **Originating Locations and Routing Policy List** page opens. In the **Originating Locations** table, select **Galway Stack**. In the **Routing Policies** table, select the Routing Policy that should be used to route the digits (e.g. **ToMESCM_PCA**).Click **Select** to save these changes.



The **Dial Pattern Details** page is displayed. Click **Commit** to save.



Since all calls from Aura clients bound for the CS 1000 will have a leading digit ‘7’ or ‘2’ inserted, the dial pattern ‘7’ or ‘2’ is also used from Communicator for Lync clients, 5-digit station calls on CS 1000 and calls to CallPilot voicemail. Follow the same steps described above to configure a dial pattern of 7 to be used to route Communication Manager calls to the CS 1000. The resultant dial pattern configuration should look as follows. The **Originating Locations Name** is the same as before (i.e. **Galway Stack)**. The **Routing Policy Name** this time should be set to route the calls to CS 1000 (i.e. **ToCS1kHA**).



Note: For 112 / 911 Emergency calls from Aura clients, it may be desired not to strip off the leading prefix of 44 from the Calling Line ID (CLID) so that Emergency Services will receive the full 7-digit CLID rather than the usual 5-digit CLID for a converged / native user. This information can help to determine that the call was made from a collaborated client rather than a prime CS 1000 phone which may aid in locating the caller in this case. To do this a second SIP Entity can be created for CS 1000 (e.g. called "CS1k_Emergency") with no adaptation applied (therefore no leading digits will be deleted from the origination address as documented in **Section 5.3**). Then a Routing Policy can be created specifically to route digits 112 or 911 to this SIP Entity where the full 7-digit incoming CLID will be presented to CS 1000 for further handling per standard emergency services. Detailed instructions and screenshots of this configuration are not shown in this document.

7. Configure Avaya Aura® Communication Manager

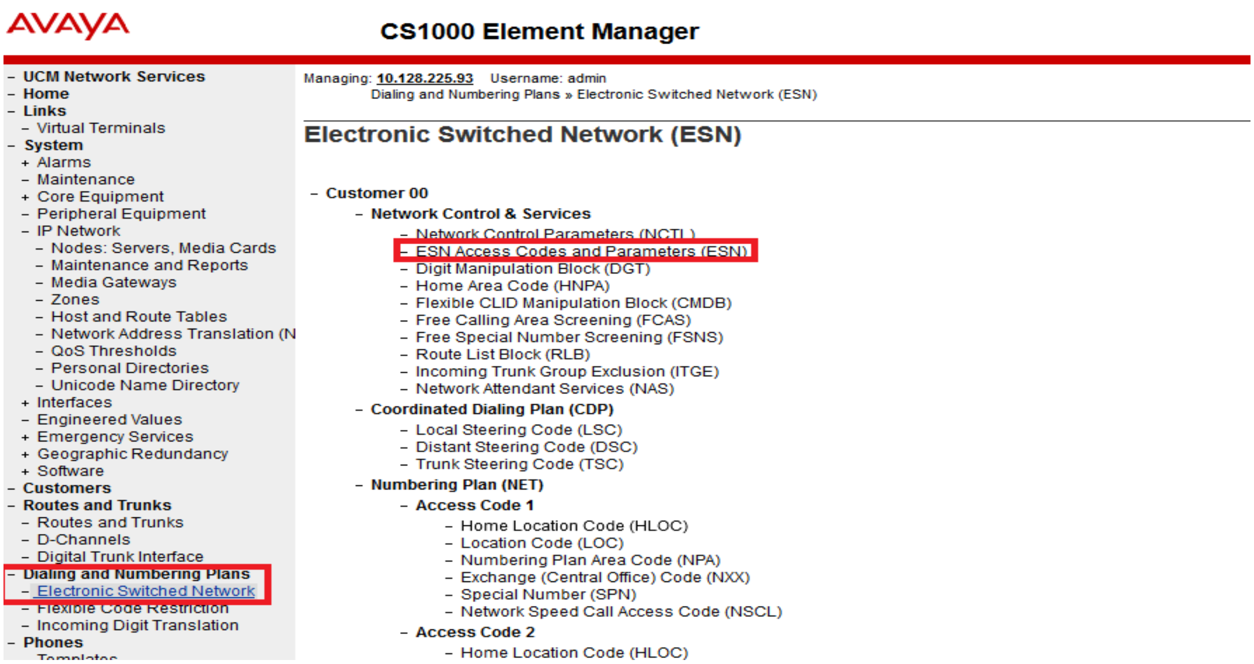
This section describes the steps required to configure Communication Manager (as an Evolution Server) to support Communicator for Lync Users with Avaya Aura® Midsize Enterprise. These instructions assume the Avaya G430 (or equivalent) Gateway is already configured on Communication Manager.

The following administration steps will be described:

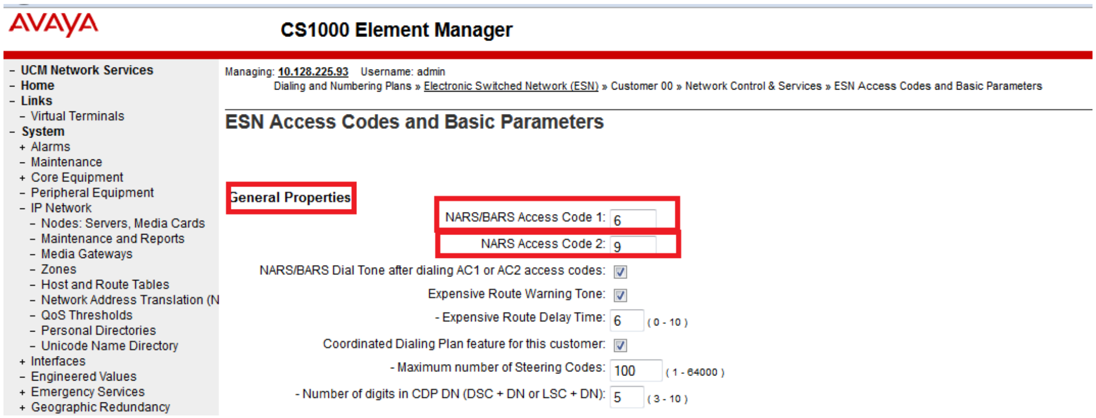
- Verify System Access codes match
- Verify IP Network Region – SIP Domain
- Configure Trunk-to-Trunk transfers
- Administer SIP Signaling Group
- Administer SIP Trunk Group
- Verify Signaling Group and Trunk Group are in-service
- Configure incoming call handling for the SIP trunk group
- Administer Private and Public Numbering Plans
- Administer Uniform Dial plan
- Administer a Route Pattern
- Administer ARS Analysis
- Administer ARS Digit Conversion

7.1. Verify System Access Codes match

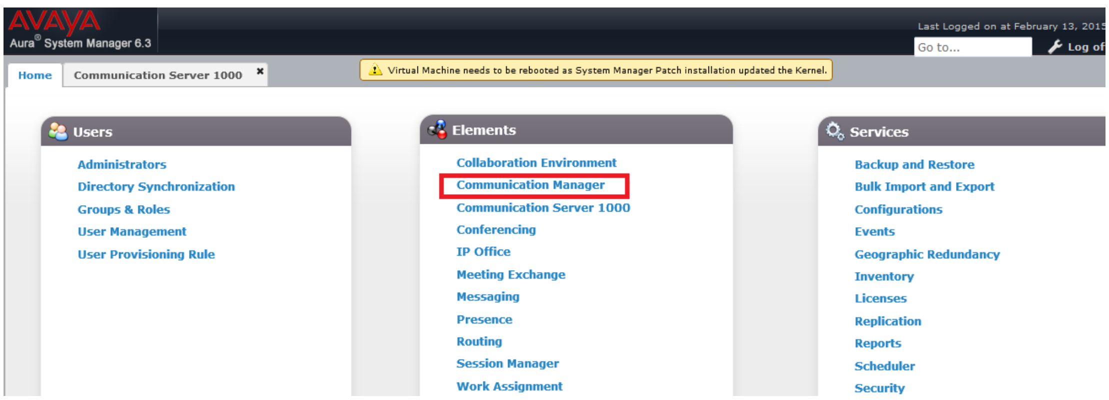
To allow users to utilize Converged dialing plans to route calls, verify Communication Manager AAR and ARS access codes match CS 1000 Access Codes. From System Manager Home Page under the Services category, navigate to UCM Services for CS 1000 management (as described earlier in **Section 4.2**).The Avaya Unified Communications Management Elements page opens in a new browser window. Under the Element Name column select one of the elements corresponding to CS 1000 in the Element Type column (e.g. EM on cs1kcores1). From the **CS 1000 Element Manager** page, expand **Dialing and Numbering Plans** on the left navigational panel and select **Electronic Switched Network**. The **Electronic Switched Network (ESN)** page appears. Click on the link for **ESN Access Codes and Parameters (ESN)**.



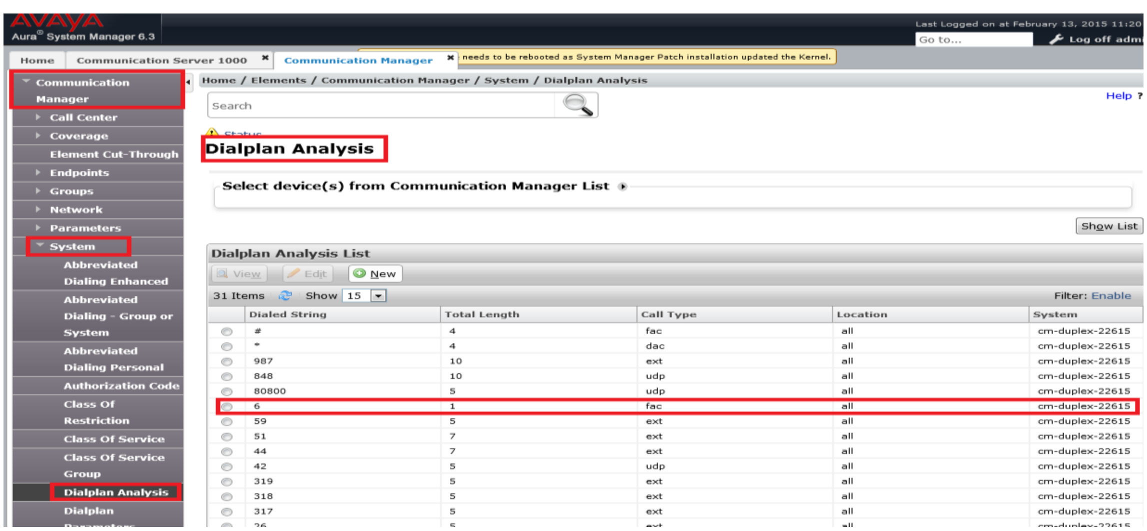
The **ESN Access Codes and Basic Parameters** page appears. Take note of the numbers used for CS 1000 **NARS/BARS Access Code 1** (e.g. **6**) and **NARS Access Code 2** (e.g. **9**).



From the main System Manager page under **Elements**, click on **Communication Manager**.

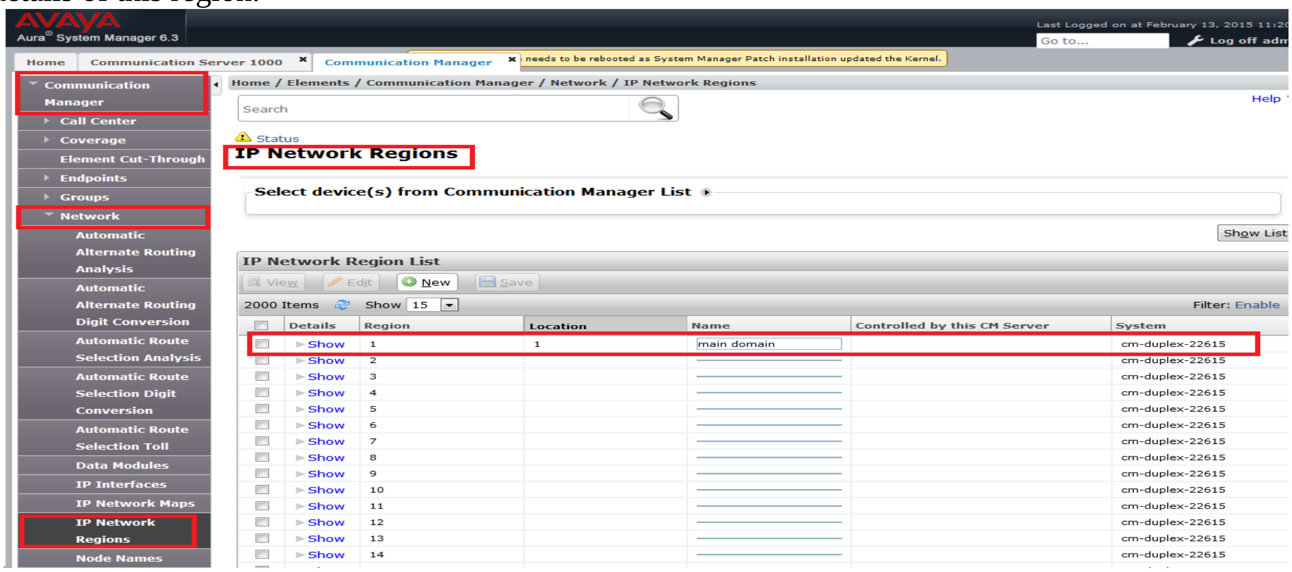


Under the **Communication Manager** list on the left hand side, expand the **System** list and select **Dialplan Analysis**. It is presumed that in the initial configuration of the Midsize Enterprise / Communication Manager, dial codes of 9 and 6 may already have been configured as they are commonly used codes. If so, they will show up on the **Dialplan Analysis List** (e.g. **9** is shown below). Verify that the **Total Length** is **1** and the **Call Type** is **fac** (this identifies the one-digit number as a Feature Access Code).



7.2. Verify IP Network Region –SIP Domain

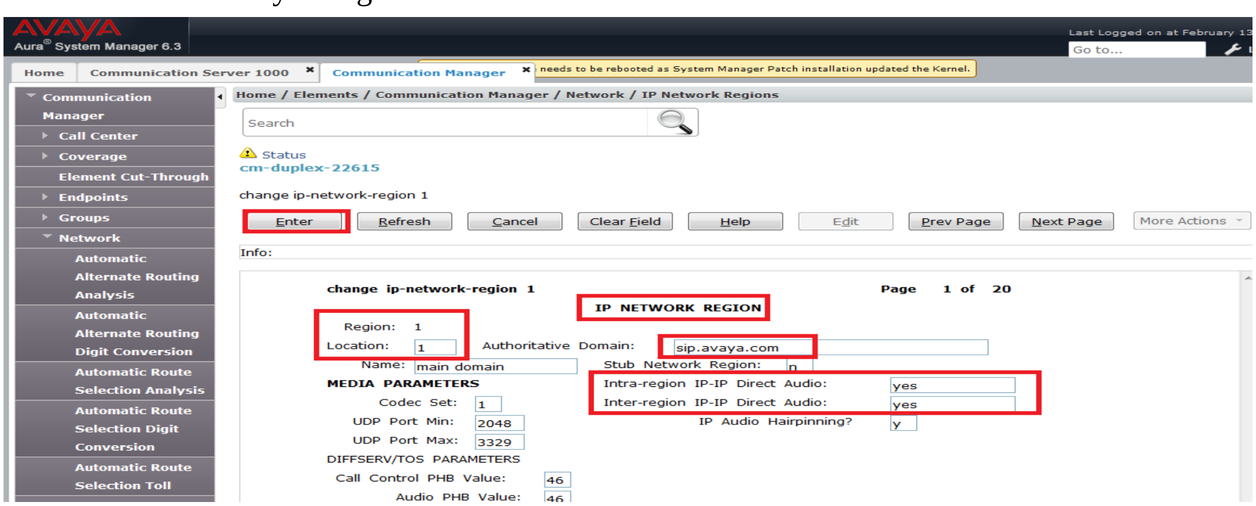
From the main System Manager page navigate to Elements and to Communication Manager. Under the **Communication Manager** list on the left hand side, expand the **Network** list and select **IP Network Regions**. From the **IP Network Region List** select Region **1** (presuming that this has already been configured on the Communication Manager). Click on **Edit** to view the details of this region.



Page 1 of the **IP NETWORK REGION** appears. Verify or enter the following values and use default values for remaining fields.

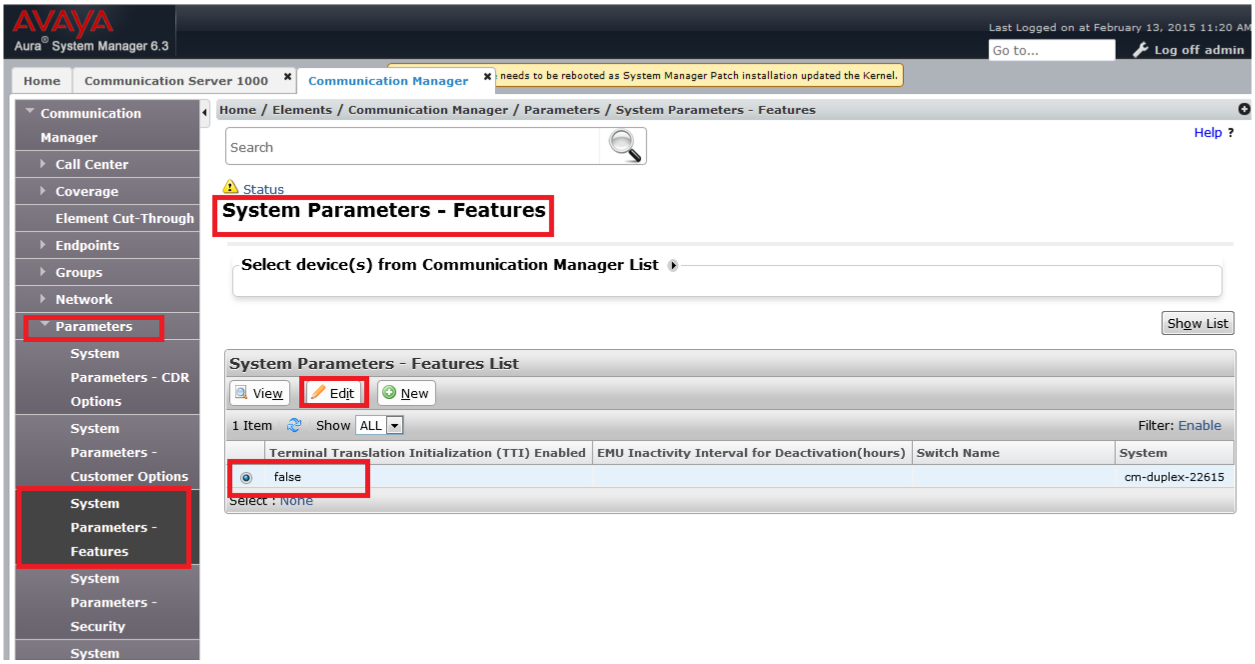
- **Authoritative Domain:** Enter the correct SIP domain for the configuration (e.g. **Svstack.com**).
- **Name:** Enter a descriptive name (e.g. **LOCAL**).
- **Codec Set:** Enter **1**
- **Intra-region IP-IP Direct Audio:** Enter **yes**
- **Inter-region IP-IP Direct Audio:** Enter **yes**

Click on **Enter** if any changes were made otherwise click on **Cancel**.

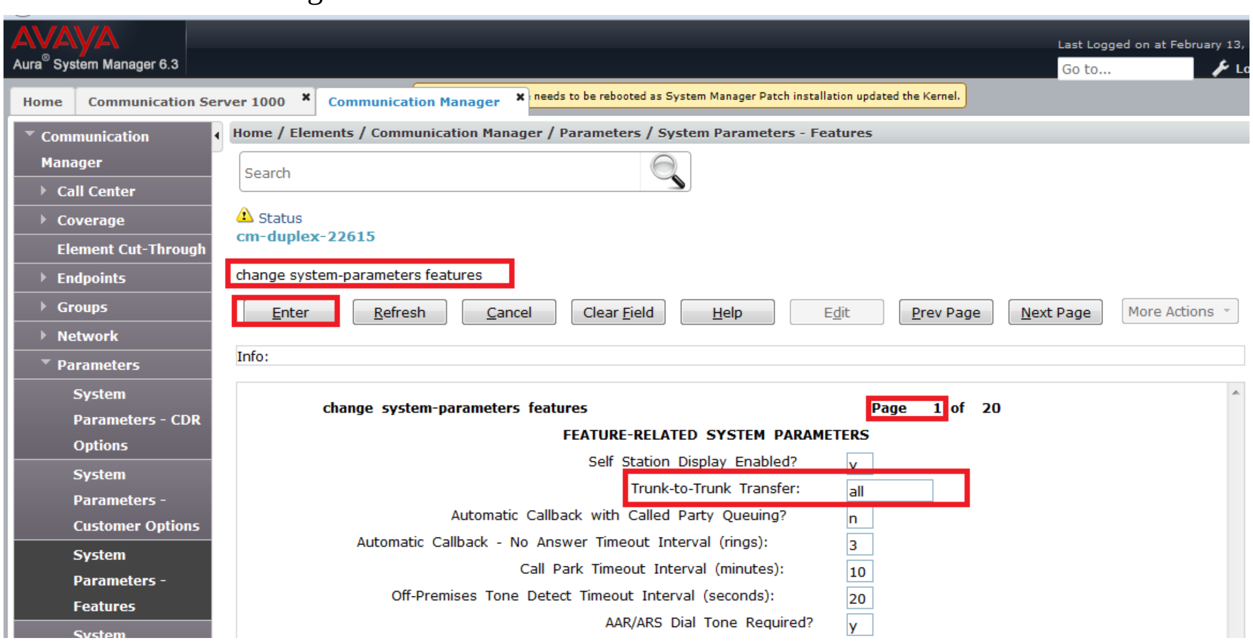


7.3. Configure Trunk-to-Trunk Transfers

From the main System Manager page navigate to Elements and to **Communication Manager**. Under the **Communication Manager** list on the left hand side, expand the **Parameters** list and select **System Parameters - Features**. Select the **System Parameters – Features List** button item (shown as **false)**, and click on **Edit**.



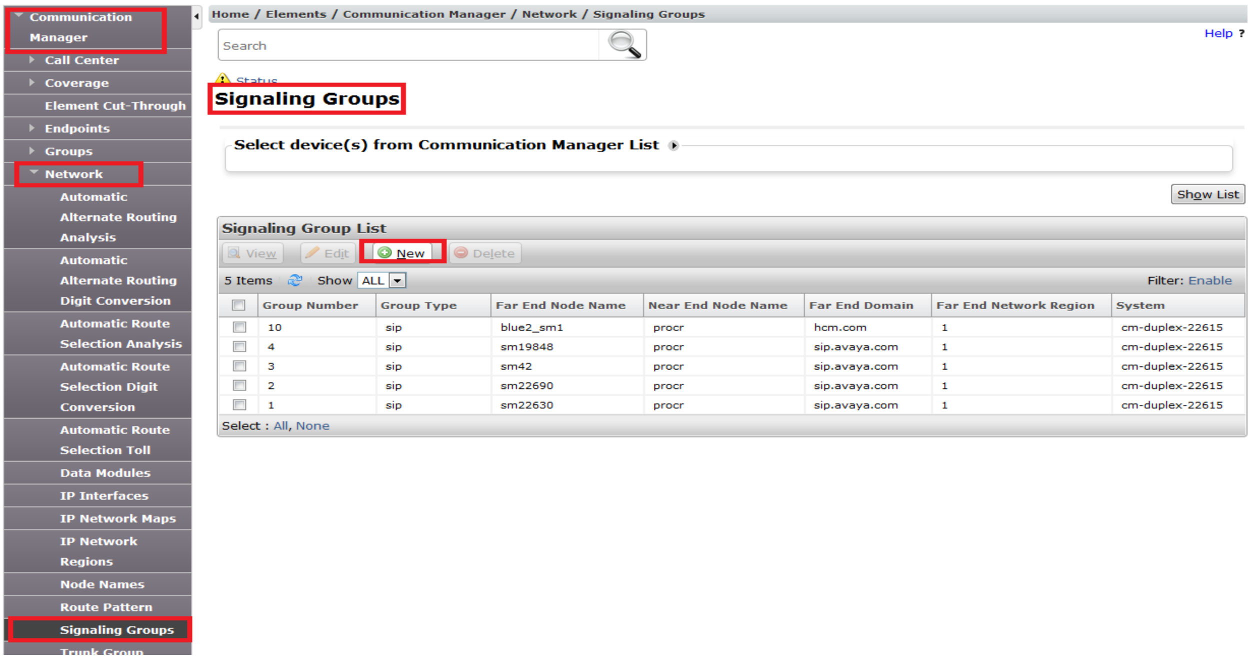
The **change system-parameters features** page appears. Enable trunk-to-trunk transfers on a system wide basis to allow an incoming call to a SIP station to be transferred to another SIP station. Set the **Trunk-to-Trunk Transfer** field on **Page 1** to **all** using the drop-down list. Press **Enter** to save the change.



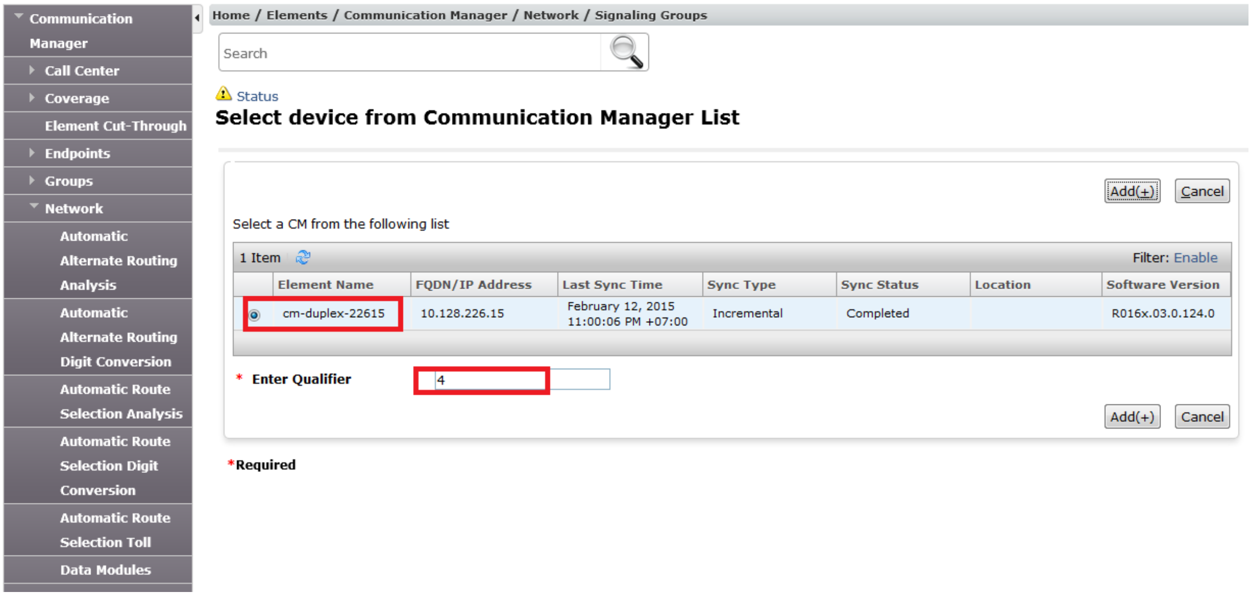
7.4. Administer SIP Signaling Group

Signaling Group 3 and an associated SIP Trunk Group 3 is presumed to have already been setup and configured for the solution with the ME installation. Signaling Group 4 and an associated SIP Trunk Group 4 will be used for all Communication Manager traffic to the CS 1000 and needs to be configured to use a different TLS port from the default.

From the main System Manager page navigate to Elements and to **Communication Manager**. Under the **Communication Manager** list on the left hand side, expand the **Network** list and select **Signaling Groups**. The **Signaling Group List** page appears. Click on **New**.



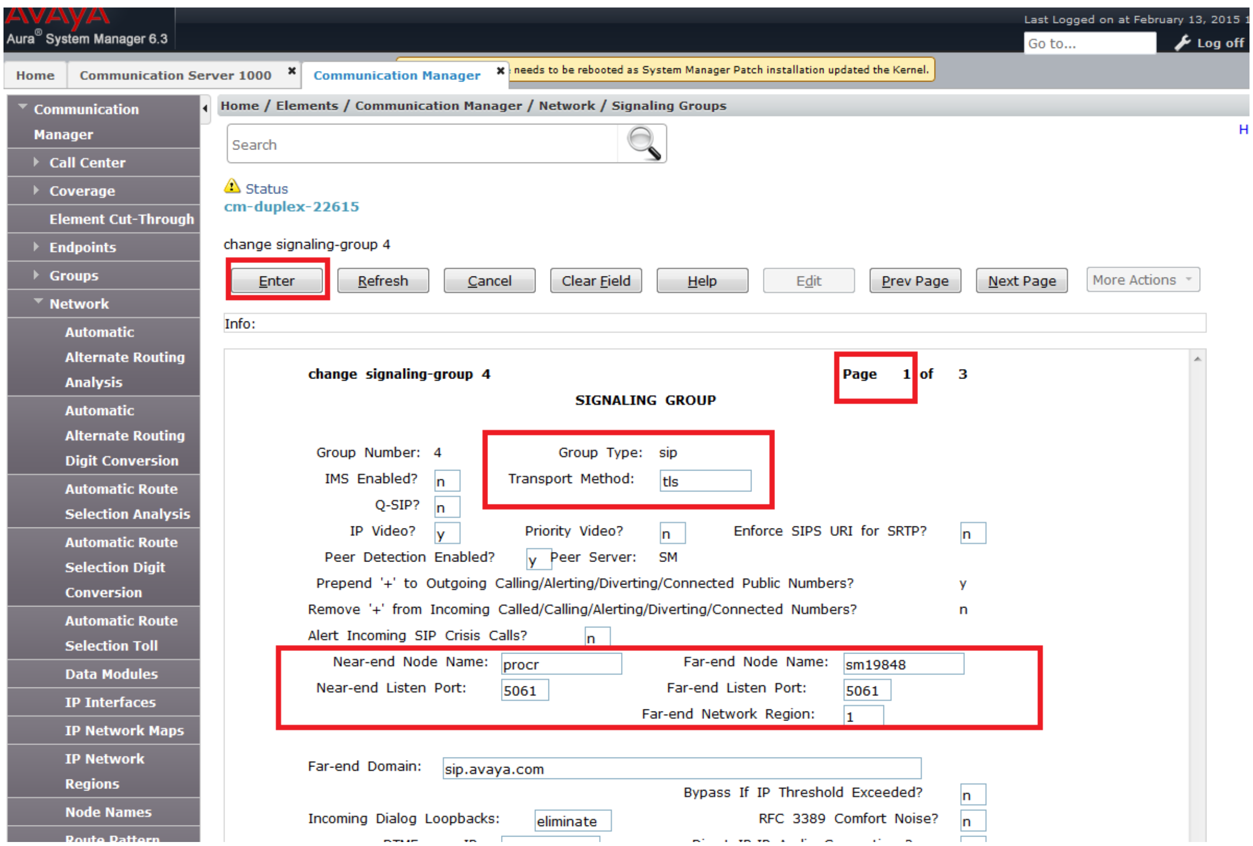
The **Select Device(s) from Communication Manager List** page appears. Select the Communication Manager (or it may be automatically selected) and in the **Enter Qualifier** field, enter **4** (this is the signaling group number) and select **Add**.



Page 1 of the **add signaling-group 4** page appears. Select or enter the following values and leave all other fields as default.

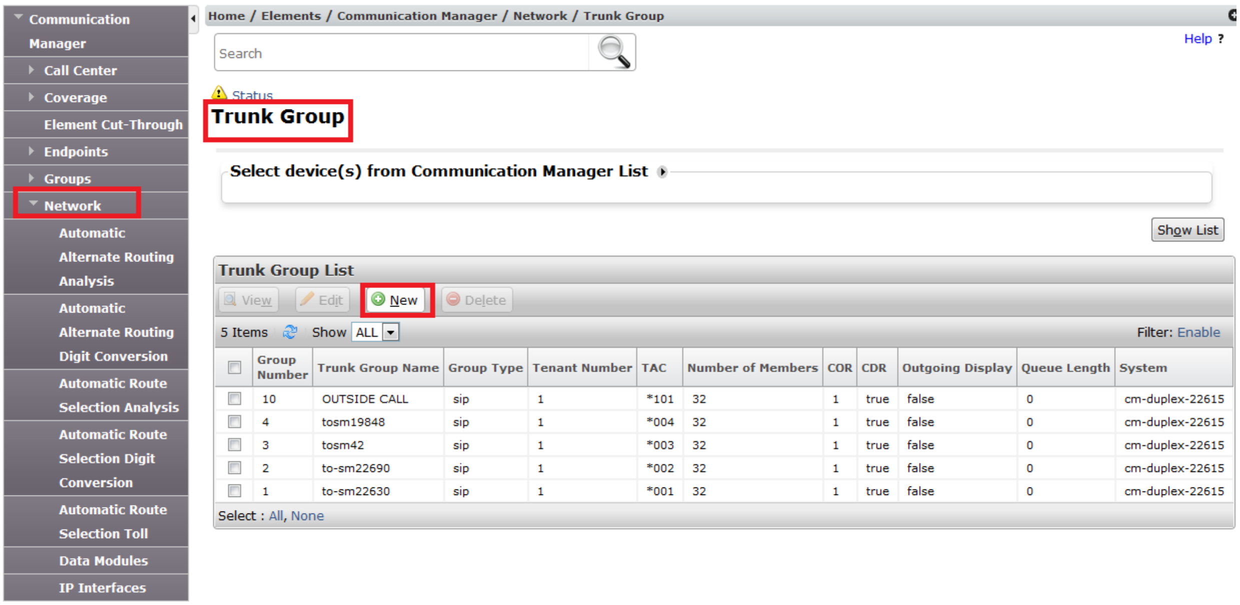
- **Group Type:** Select **SIP** from the drop-down list.
- **Transport Method: TLS** may be selected by default (preferred choice).
- **Near-End Node Name:** Select **procr** from the drop-down list (this name is created during the ME server install).
- **Far-End Node Name:** Select **SM** from the drop-down list (this name is created during the ME server install).
- **Near-End Listen Port:** Enter**5062** as a different port number than the default used by signaling group 3.
- **Far-End Listen Port:** Enter **5062** as a different port number than the default used by signaling group 3.
- **Far-End Network Region:** Enter**1** in this field.
- **Far-end Domain:** Enter the SIP domain for the configuration (e.g. **svlstack.com**).

Click on **Enter** to save the new signaling group.

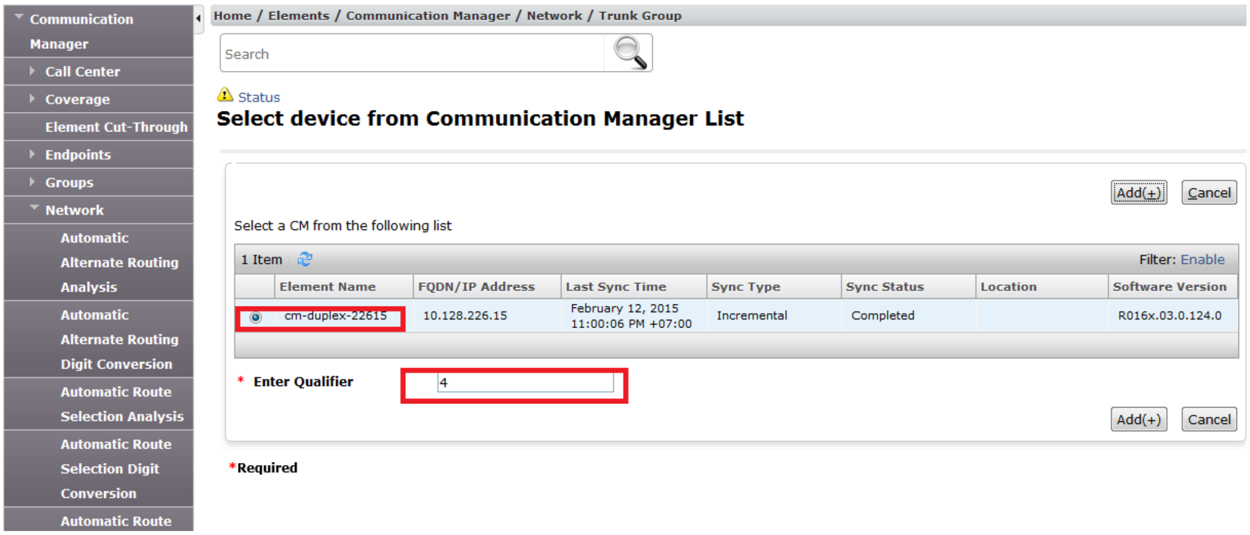


7.5. Administer SIP Trunk Group

Trunk Group 4 will use signaling group 4 created in **Section 7.4** and will be used for all traffic to theCS 1000.From the main System Manager page navigate to Elements and to Communication Manager. Under the **Communication Manager** list on the left hand side, expand the **Network** list and select **Trunk Group**. The **Trunk Group List** page appears. Click on **New**.



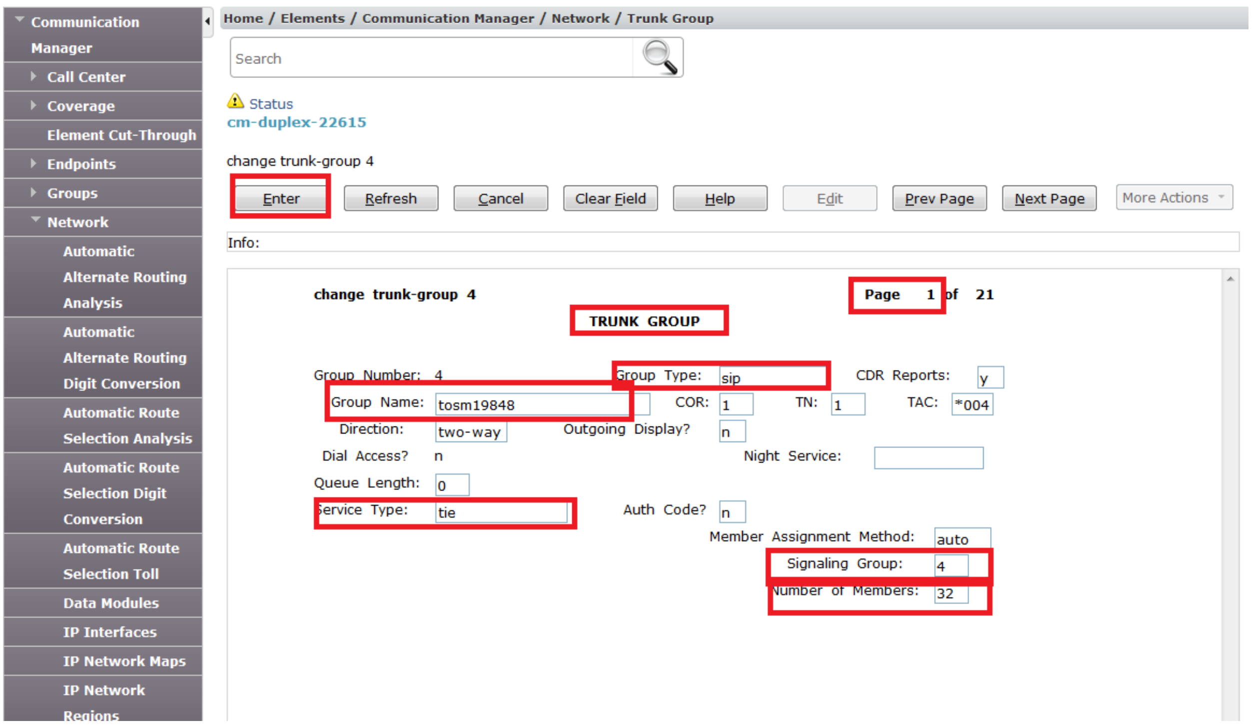
The **Select Device(s) from Communication Manager List** page appears. Select the CM (or it may be automatically selected) and in the **Enter Qualifier** field, enter **4** (this is the trunk group number) and select **Add**.



Page 1 of the **add trunk-group 4** page appears. Select or enter the following values and leave all other fields as default.

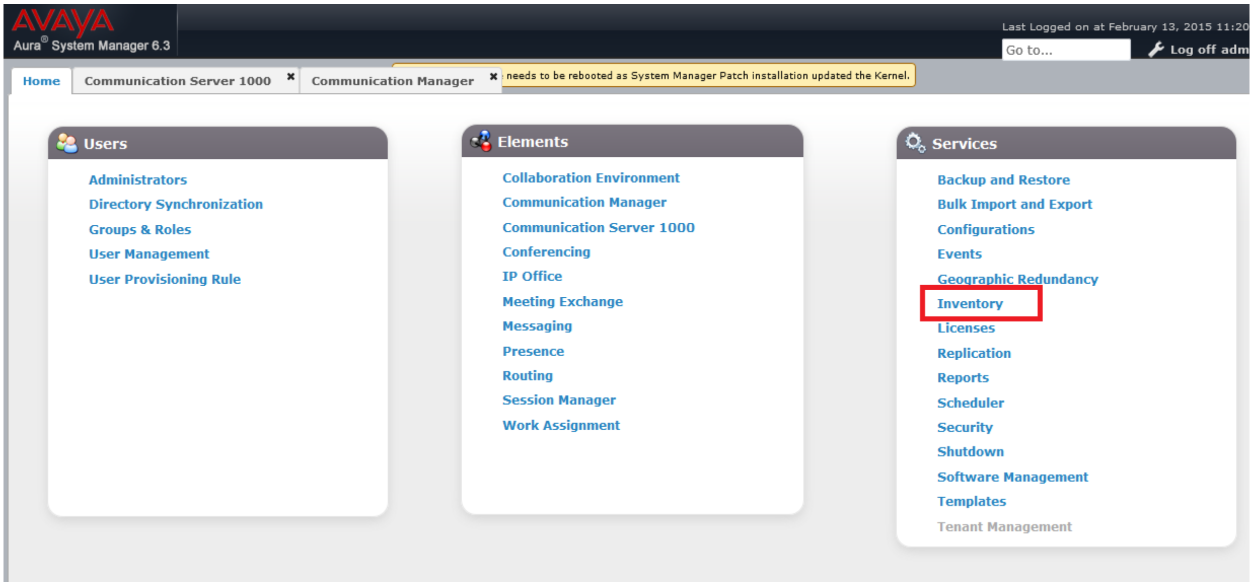
- **Group Type:** Select **SIP** from the drop-down list.
- **Group Name:** Enter a description for the trunk group (e.g. **SIP Trunk CS1k**).
- **TAC:** Enter ***04** as the trunk access code.
- **Service Type:** Enter **tie** from the drop-down list.
- **Member Assignment Method:** Select **auto** from the drop-down list.
- **Signaling Group:** Enter **4** as the signaling group created in **Section 6.4**.
- **Number of Members:** Enter **100**.

Click on **Enter** to save the new signaling group

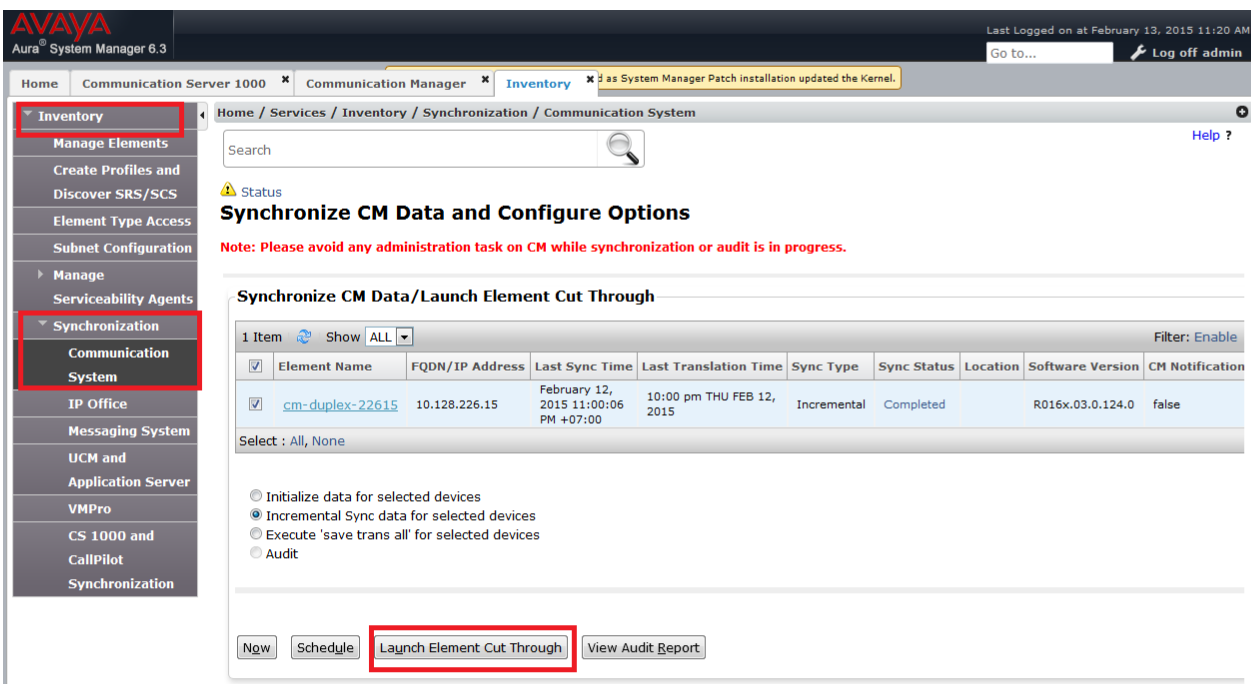


7.6. Verify Signaling Group and Trunk Group are in-service

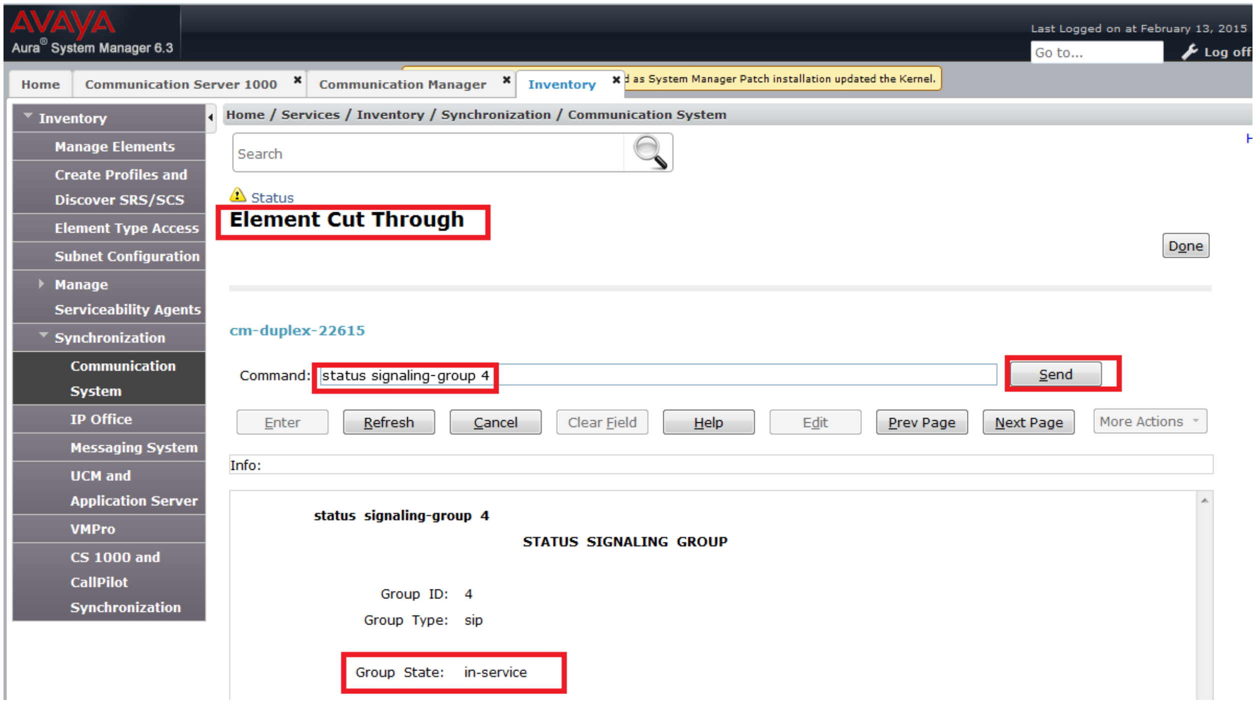
After Signaling Group 4 and Trunk Group 4 have been added, ensure that both are up and in-service. From the main System Manager page under **Elements**, click on **Inventory**.



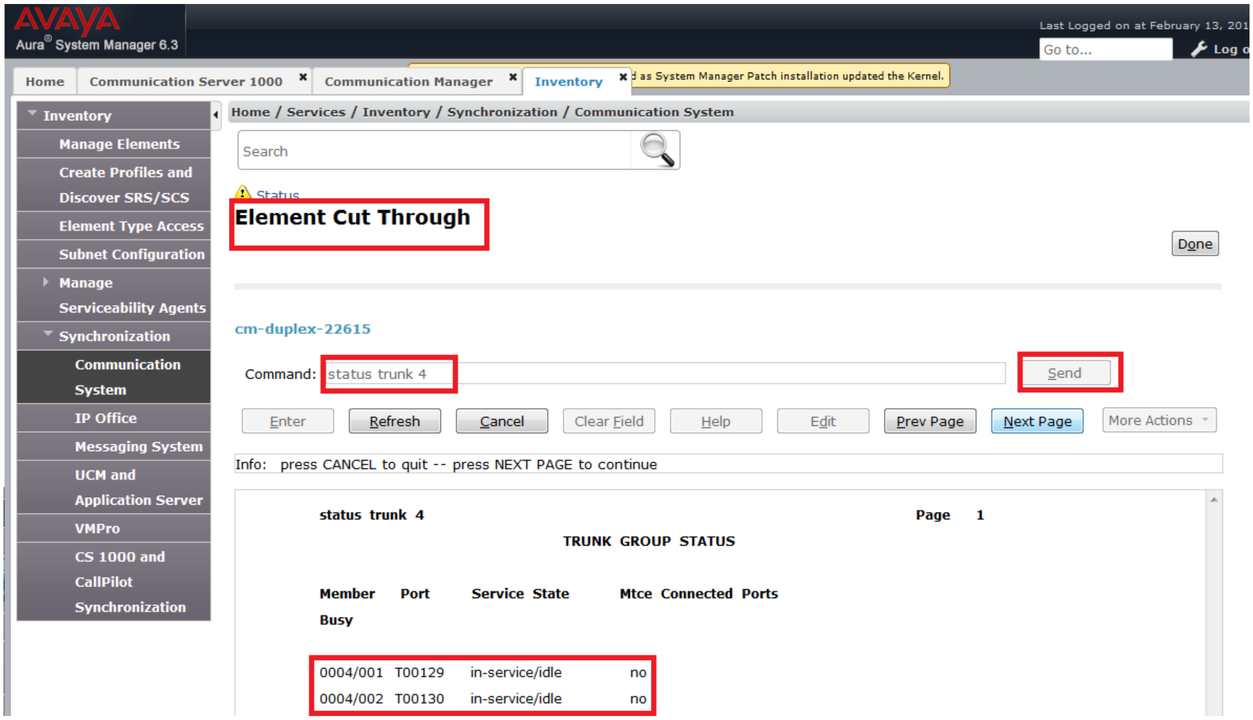
Under the **Inventory** list on the left hand side, expand the **Synchronization** list and select**Communication System**. Click on **Launch Element Cut Through**.



The **Element Cut Through** window appears. In the **Command** box, enter **status signaling-group 4** and click on **Send**. Verify that the **Group State** shows **in-service**.



In the **Command** box, enter **status trunk 4** and click on **Send**. Verify that the **Service State** shows **in-service/idle** and **Mtce Busy** is **no** for all 100 members (use **Next Page** to verify other members).



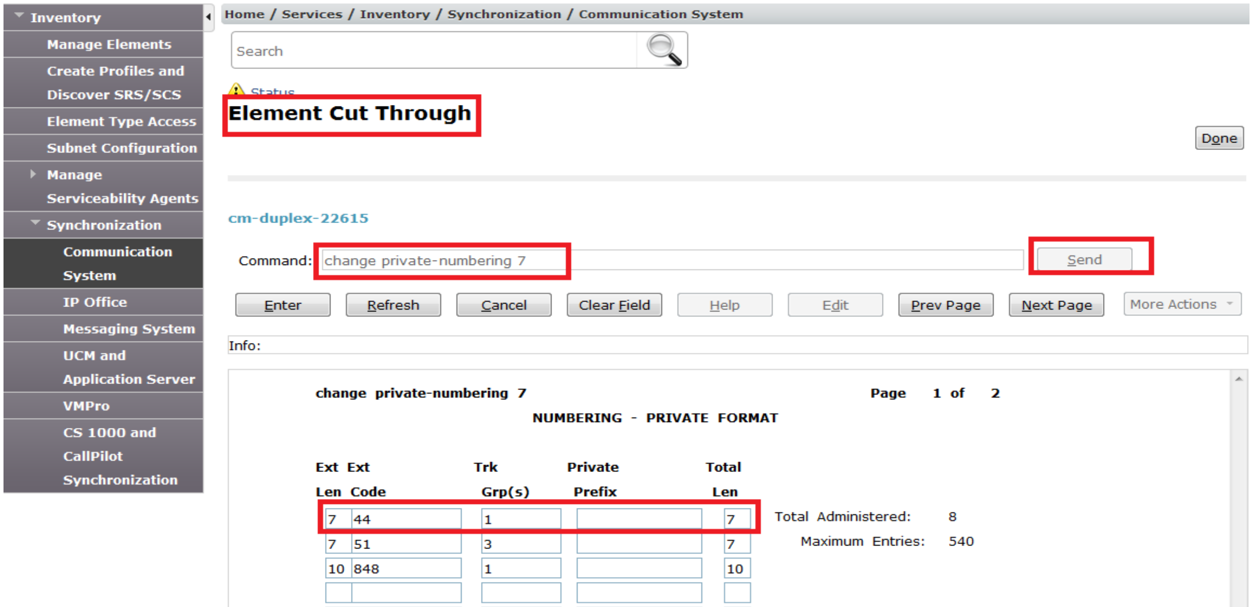
7.7. Administer Private Numbering Plan

The full extension numbers used for the Aura clients registered to Session Manager must be added to the private numbering table on Communication Manager. For the reference configuration, private numbering was used and all extension numbers were unique within the private network.

Using the same **Element Cut Through** window used in the previous **Section 7.6**, in the **Command** box, enter a command of the format "change private-numbering n", where n is the length of the private number (e.g. 7). For the sample configuration where all collaborated endpoint DNs were 7 digits long, the command is **change private-numbering 7**. Click on the **Send** button. Fill in the indicated fields as shown below.

- **Ext Len**: Enter the length of the collaborated sets extension numbers (e.g. **7**).
- **Ext Code**: Enter the leading digit(s) for the collaborated sets extension numbers (e.g. **44** was used as a prefix for all collaborated endpoints).
- **TrkGrp(s):** Enter**3** as the trunk group.
- **Total Length:** Enter **7** since a private prefix was not defined.

Click on **Enter** to accept the change.


Inventory
Manage Elements
Create Profiles and Discover SRS/SCS
Element Type Access
Subnet Configuration
Manage Serviceability Agents
Synchronization
Communication System
IP Office
Messaging System
UCM and Application Server
VMPro
CS 1000 and CallPilot Synchronization
Home / Services / Inventory / Synchronization / Communication System
Search
Status
Element Cut Through
Done
cm-duplex-22615
Command: change private-numbering 7
Send
Enter
Refresh
Cancel
Clear Field
Help
Edit
Prev Page
Next Page
More Actions
Info:
change private-numbering 7
Page 1 of 2
NUMBERING - PRIVATE FORMAT
Ext Len
Ext Code
Trk Grp(s)
Private Prefix
Total Len
7 44 1 7
7 51 3 7
10 848 1 10
Total Administered: 8
Maximum Entries: 540

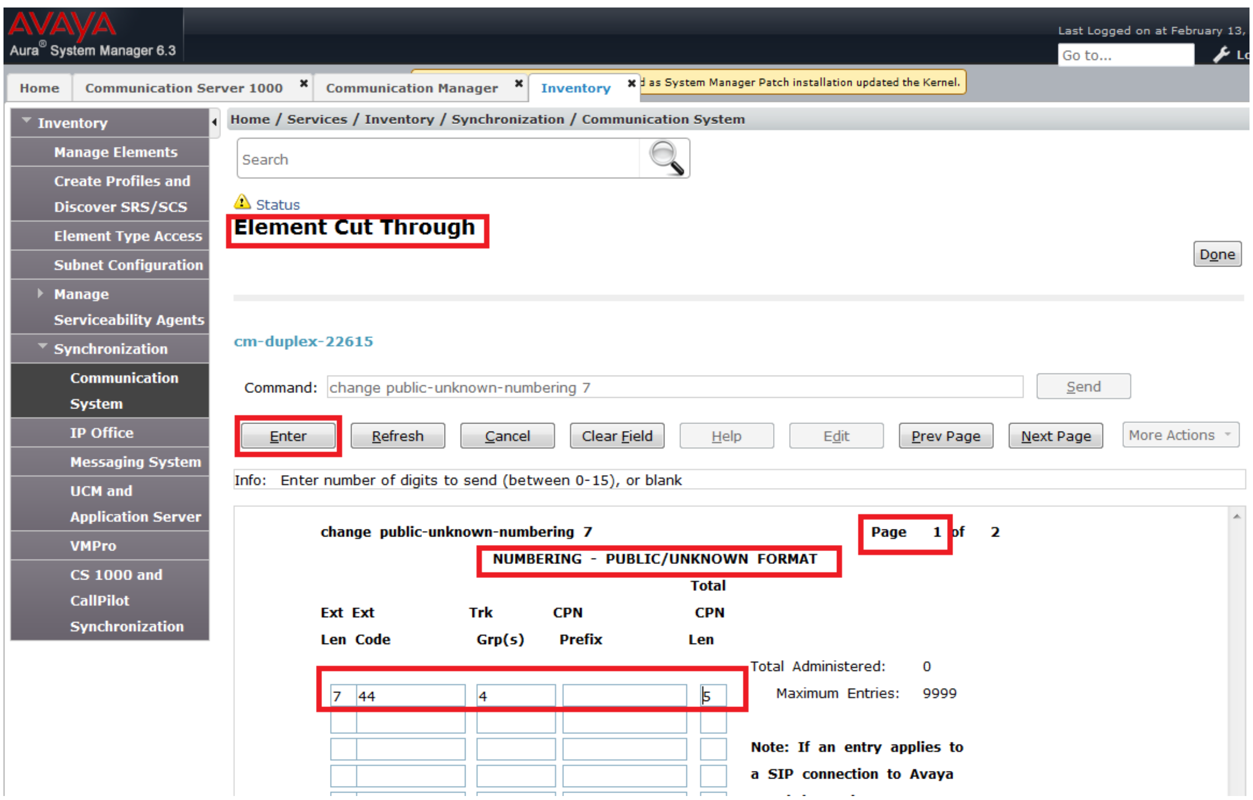
7.8. Administer Public Numbering Plan

The full extension numbers used for the Aura clients registered to Session Manager must be added to the public numbering table on Communication Manager. For the reference configuration, public numbering was used and all extension numbers were unique within the public network. These settings will ensure that a 7-digit Collaborated SIP station will show a 5-digit Calling Party Number (CPN) on outbound calls. For example, SIP extension 4470015 will show a Calling Party Number of 70015.

Using the same **Element Cut Through** window used in the previous **Section 7.6**, in the **Command** box, enter a command of the format "change public-unknown-numbering n", where n is the length of the public number (e.g. 7). For the sample configuration, the command used was **change public-unknown-numbering 7**. Click on the **Send** button. Fill in the indicated fields as shown below.

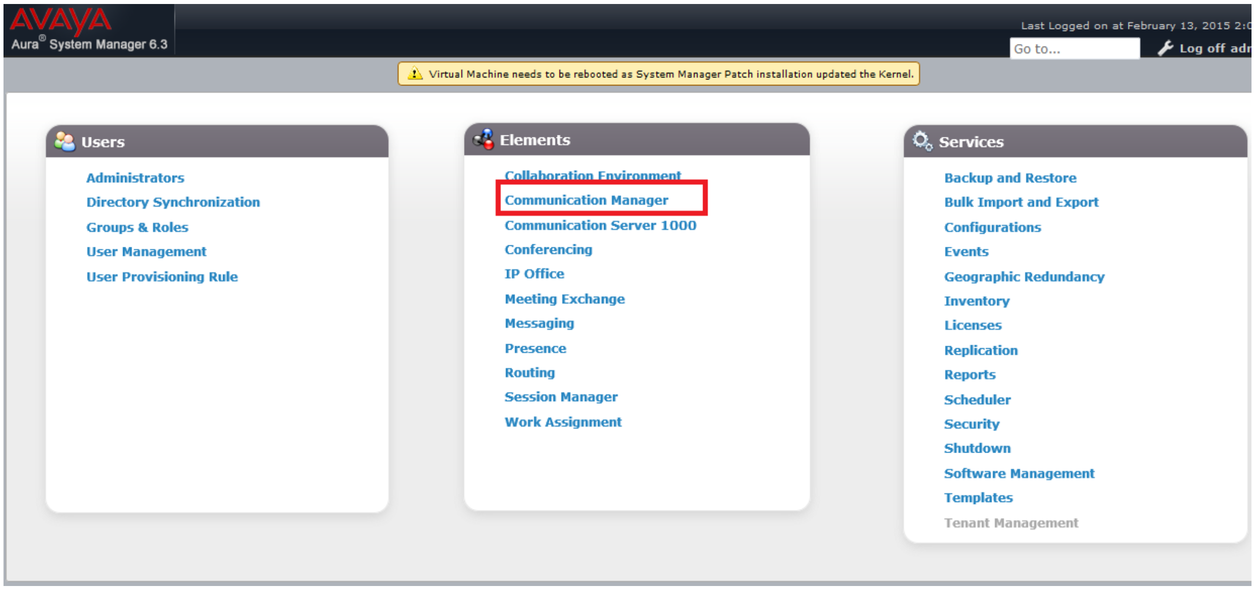
- **Ext Len**: Enter the length of the collaborated sets extension numbers (e.g. **7**).
- **Ext Code**: Enter the leading digit(s) for the collaborated sets extension numbers (e.g. **44** was used as a prefix for all collaborated SIP endpoints).
- **TrkGrp(s):** Enter **4** as the trunk group in this case.
- **Total Length:** Enter **5** since a private prefix was not defined.

Click on **Enter** to accept the change.



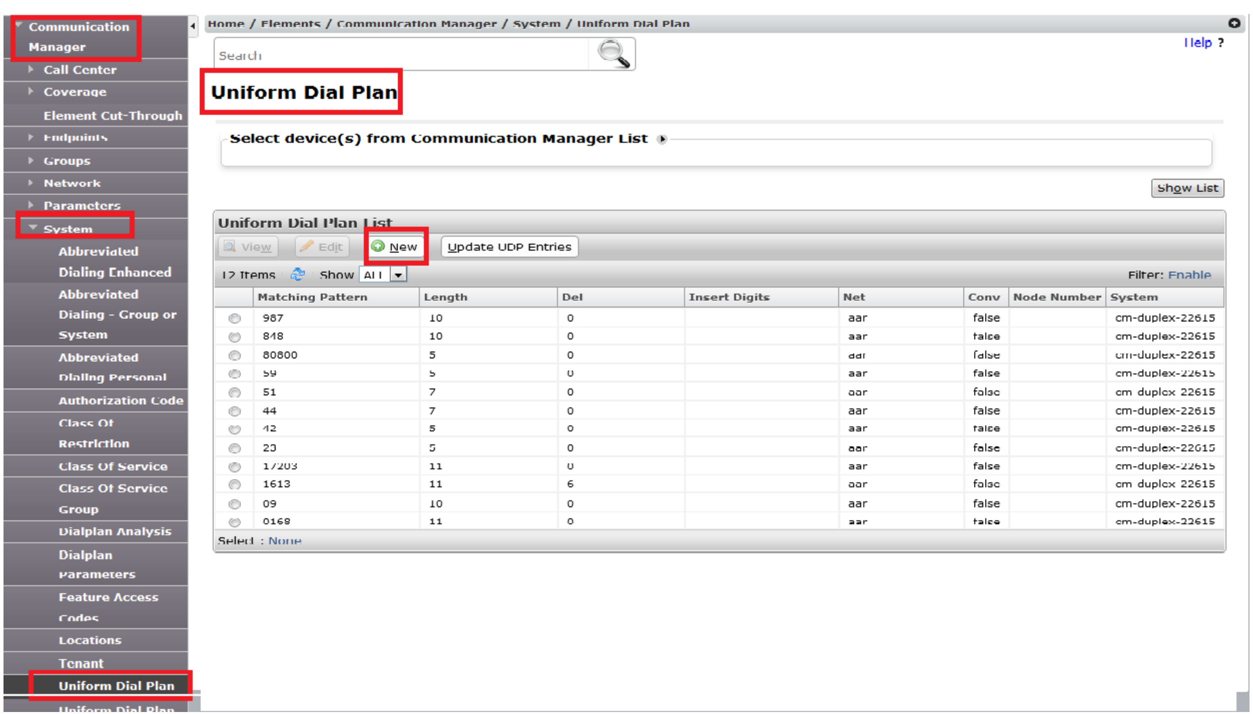
7.9. Administer Uniform Dial Plan

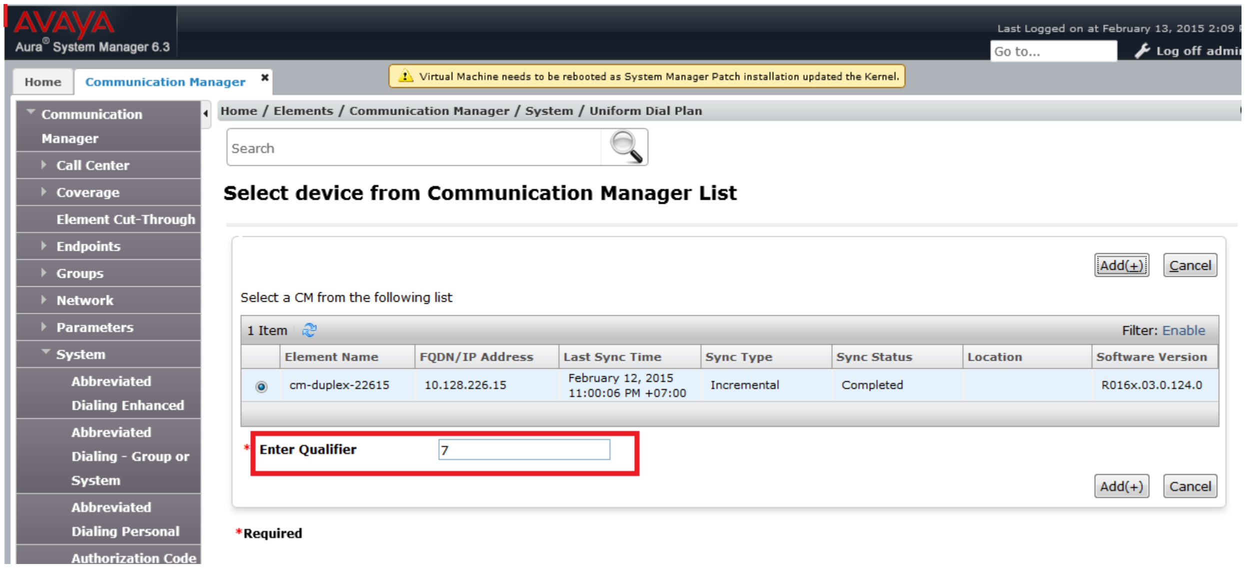
From the main System Manager page under **Elements**, click on **Communication Manager**.



Under the **Communication Manager** list on the left hand side, expand the **System** list and select **Uniform Dial Plan**. Click on **New**.

As all extensions on the CS 1000 in the sample configuration begin with the digit 7 or 2, enter **7** into the **Enter Qualifier** box. (Do the same for ‘2’)

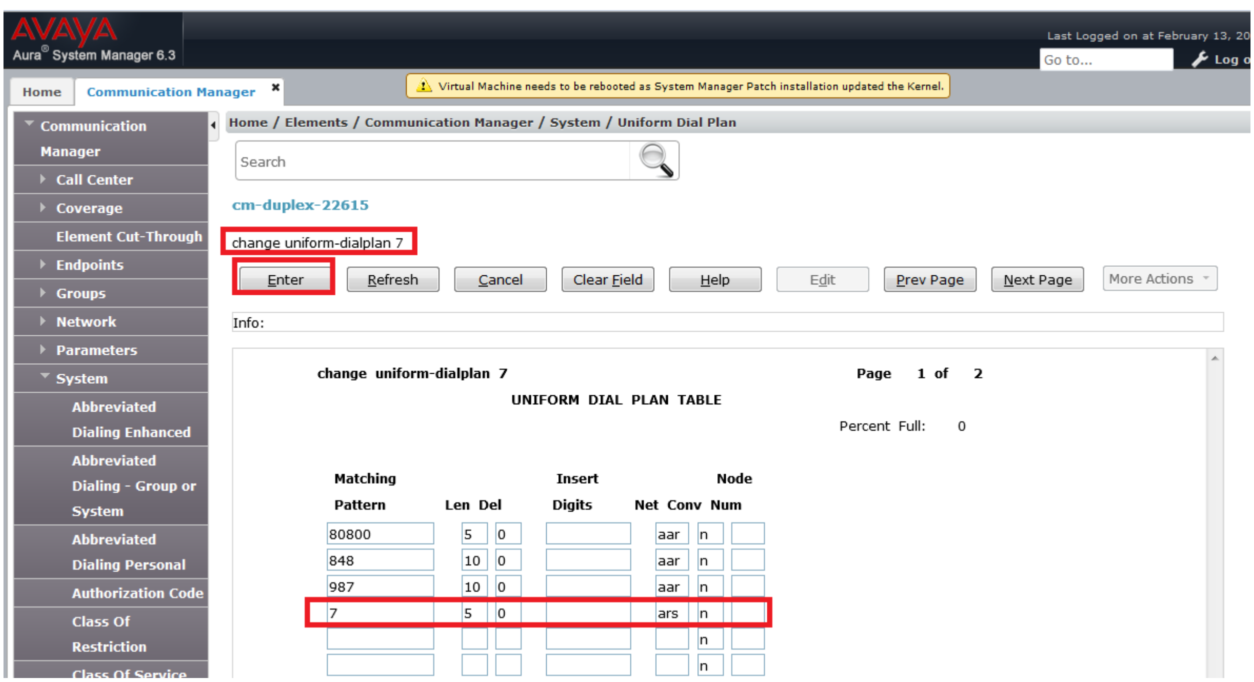




The **change uniform-dialplan 7** window appears. In the reference configuration, **7xxxx** was used as the uniform dialing plan for Converged Users dialing across CS 1000/ Aura solution. Fill in the indicated fields as shown below and use default values for remaining fields.

- **Matching Pattern** Enter a digit pattern to represent the prime extension number for all user endpoints in the solution (e.g. enter **7**).
- **Len** Enter extension length (e.g. **5**).
- **Del** Enter number of digits to delete (e.g. **0**).
- **Net** Enter **ars** from the drop-down list.

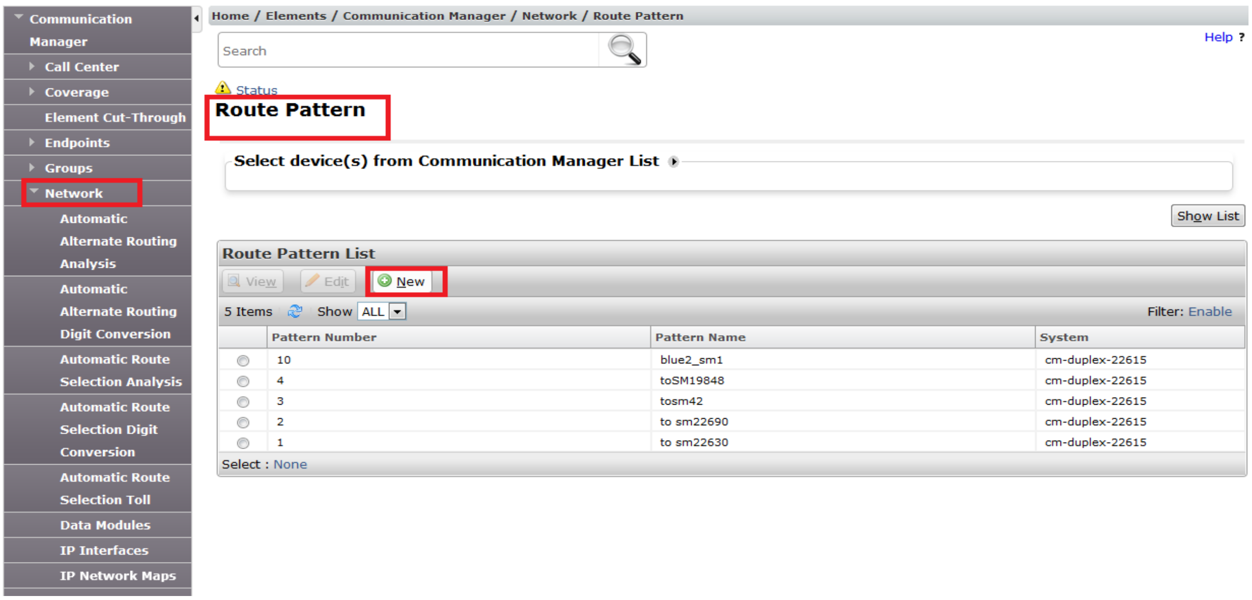
Click on **Enter** to accept the change.



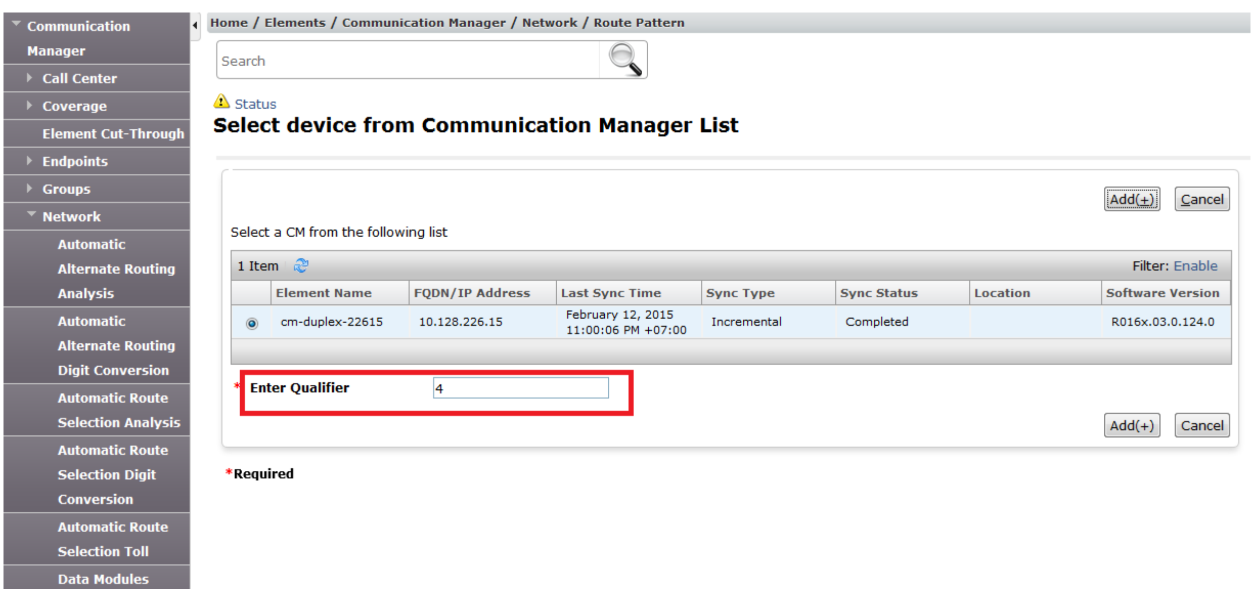
7.10. Administer Route Pattern

This section describes the configuration of the Route Pattern used in Communication Manager for the routing of calls to CS 1000. All calls from ARS will use this route pattern 4. Route Pattern 3 is created during the ME server installation and is dedicated for the IMS signaling required by the Aura clients. Using a separate route pattern (route pattern 4 in this sample configuration) allows for digit manipulation on enterprise calls without affecting the IMS traffic that is using Route Pattern3.

From the main System Manager page navigate to Elements and to Communication Manager. Under the **Communication Manager** list on the left hand side, expand the **Network** list and select **Route Pattern**. The **Route Pattern List** page appears. Click on **New**.



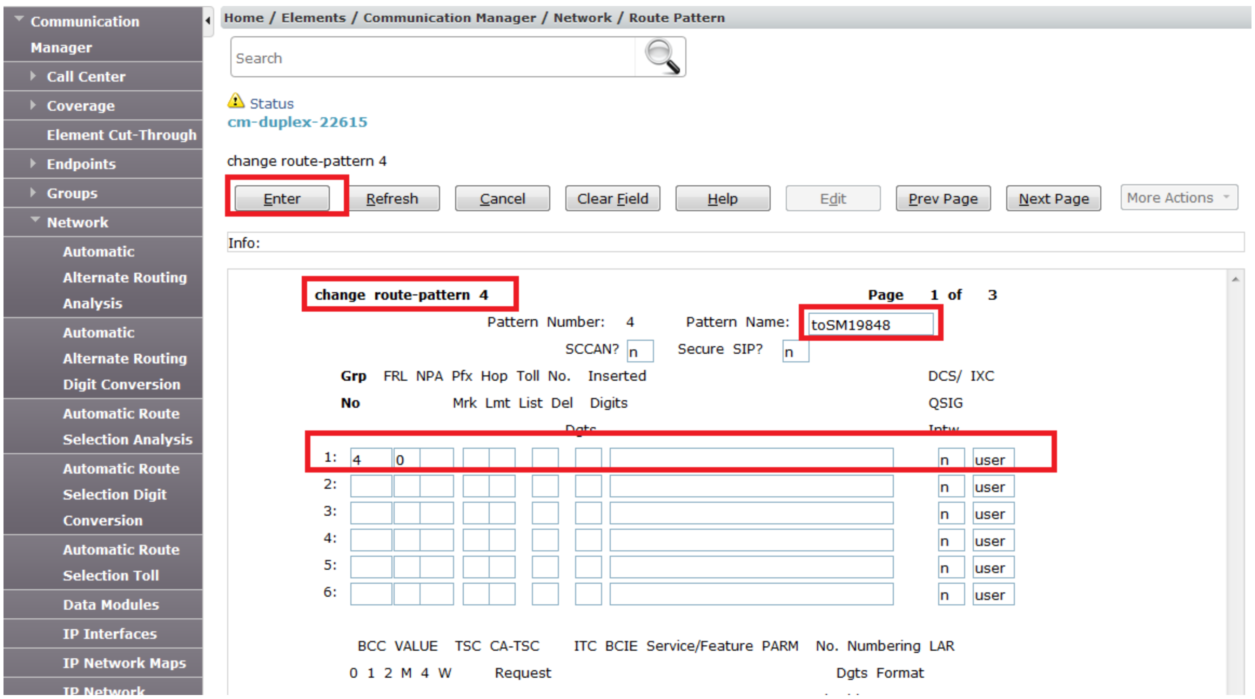
Enter the route pattern number **4** into the **Enter Qualifier** box and click on **Add**.



In the **change route-pattern 4**page which appears, enter the following values and use default values for remaining fields.

- **PatternName** Enter a description of the route-pattern (e.g. **ToCS1K**).
- **Grp No** Enter **4** as the trunk group number to be used for this route.
- **FRL** Enter **0** as the minimal facility access code restriction value.
- **Inserted Digits** Enter **1** which is the lead routing digit that Session Manager will use to point calls to CS 1000.

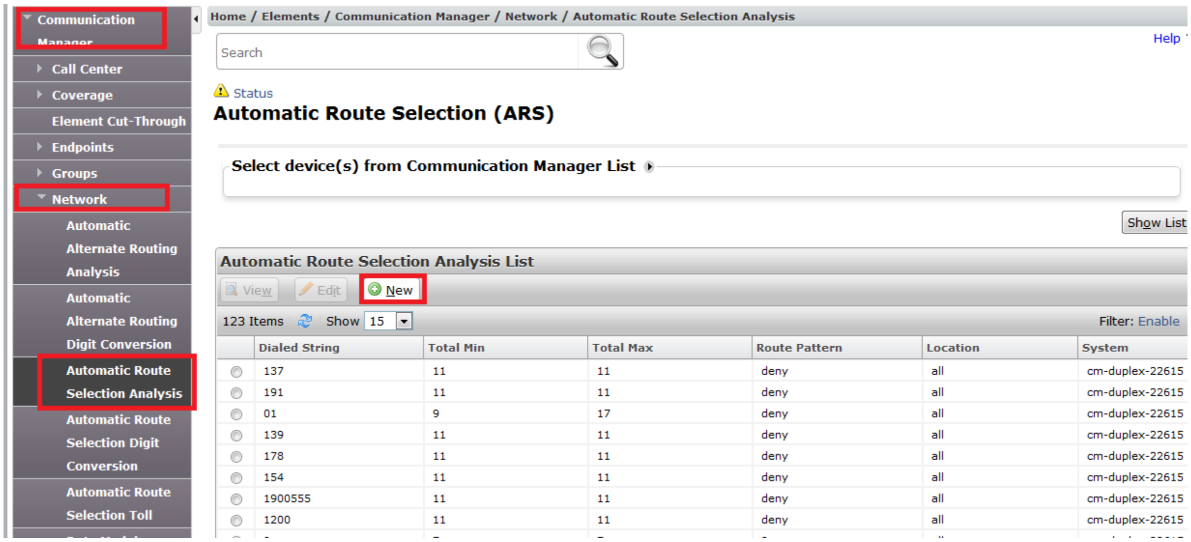
Click on **Enter** to accept the changes.



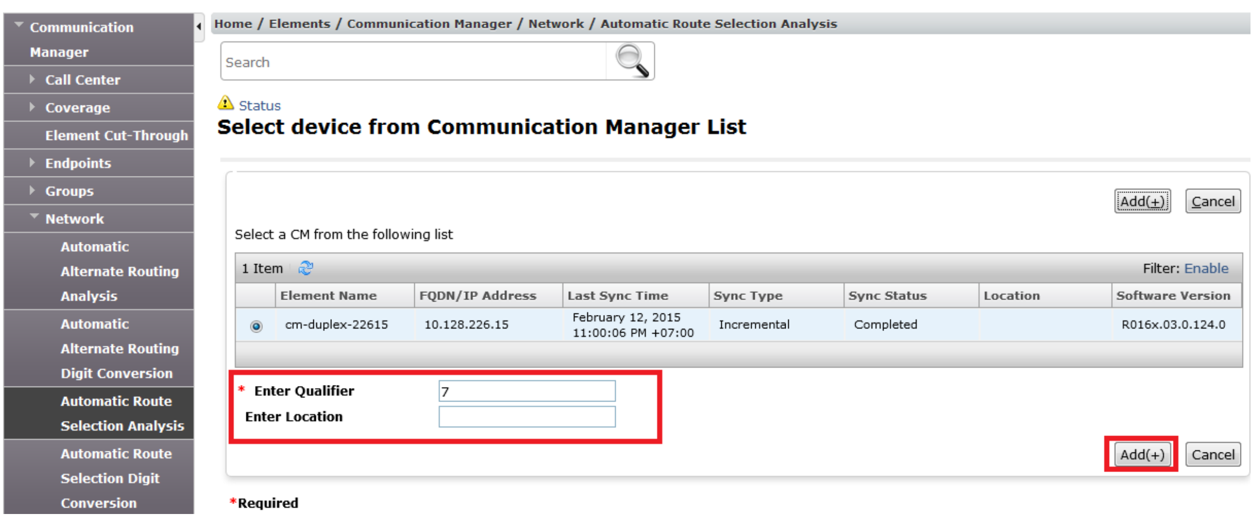
7.11. Administer ARS Analysis

This section details the configuration of the ARS pattern used in the reference configuration for routing calls between Aura clients and CS 1000 stations. All dialed numbers identified on the ARS table will be sent to Session Manager and routed to the CS 1000 via Route pattern 4created in **section 6.11**.

From the main System Manager page navigate to Elements and to Communication Manager. Under the **Communication Manager** list on the left hand side, expand the **Network** list and select **Automatic Route Selection Analysis**. The **Automatic Route Selection (ARS)** page appears. Click on **New**.



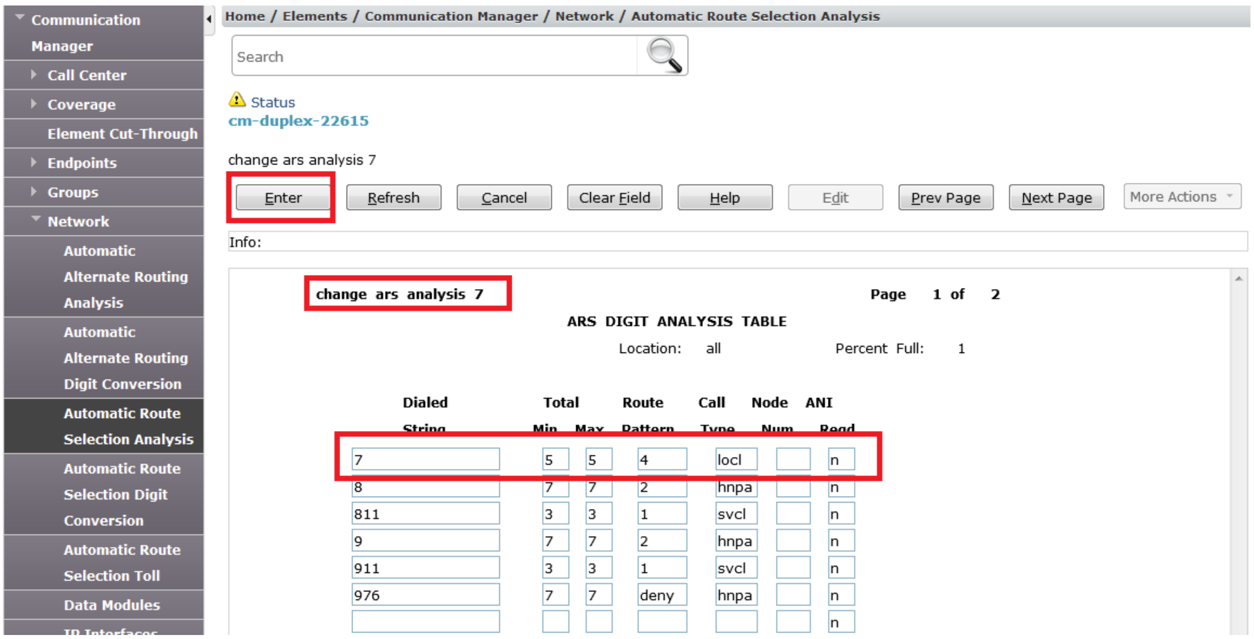
In the **Enter Qualifier** box, enter the leading digit to add e.g. in the sample configuration, all set DNs on CS 1000 start with digit 7 or 2 while all stations on Aura begin with digits 44. So in the first pass, enter **7** and in the **Enter Location** box, leave it blank to include all locations. Click on **Add**. (Do the same for '2')



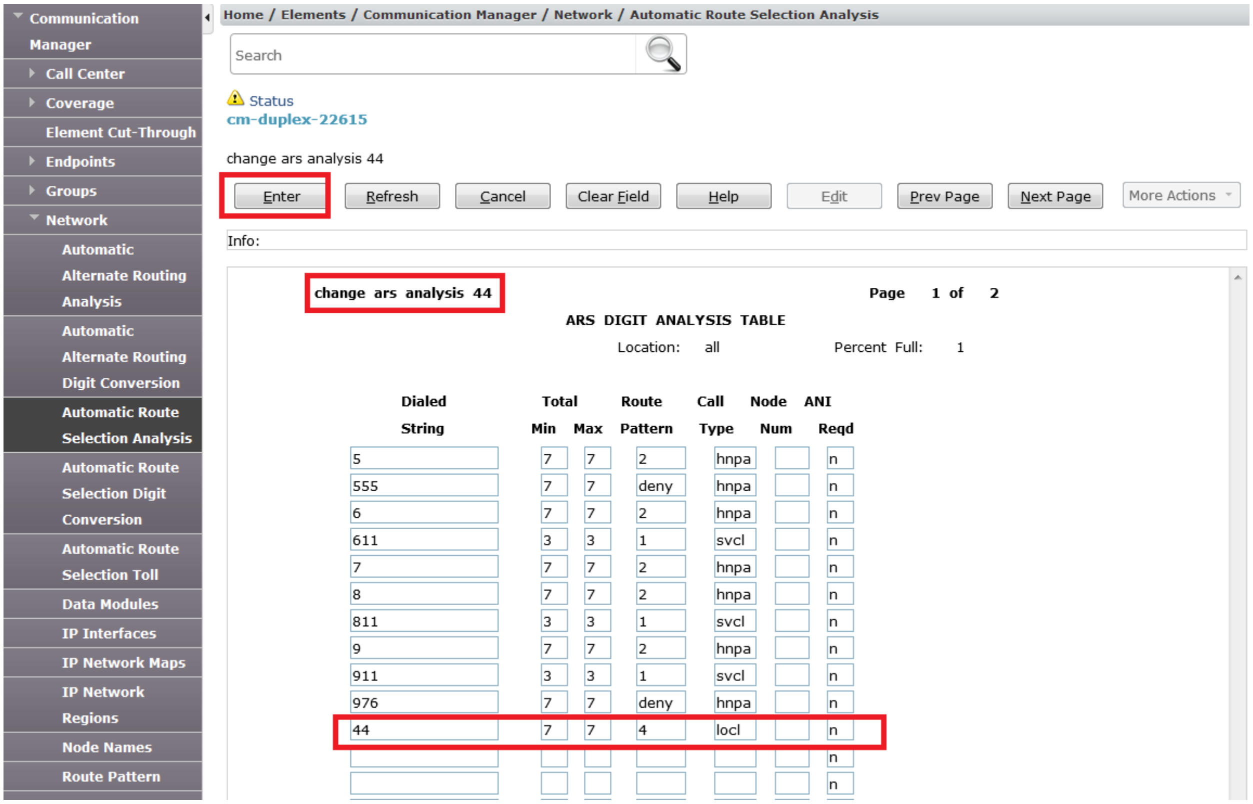
The **change ars analysis 7** page appears. Enter the following values per the sample configuration example:

- **Dialed String** Enter leading digit(s) of extension numbers (e.g. enter **7**).
- **Min** Enter the minimum number of digits that must be dialed (e.g. **5)**.
- **Max** Enter the maximum number of digits that may be dialed (e.g. **5).**
- **Route Pattern** Enter the Route Pattern for the call (e.g. **4**).
- **Call Type** Enter **locl** for a Local call.

Click on **Enter** to complete the change.



Repeat the same steps to add a Dialed String of 44 for calls to collaborated endpoints (except this time, use a Min / Max value of 7). The completed ARS table should look like the following example.



8. User Management

This section describes the details for configuring Converged and Native users across the CS 1000 and the Aura using Element Manager and System Manager User Management.
The Communication Address and Profile Extension number defined for Session Manager and Communication Manager in the sample configuration is a seven-digit number which is identical to the CS 1000 primary Directory Number plus the route prefix (e.g.**44**70xxx). The following assumes Midsize Enterprise template is configured as the Primary Security Server for the Unified Communications Management application and CS 1000 is registered as a member of the System Manager Security framework.

In the sample configuration, a user with CS 1000 DN of 20408 already exists and a Communicator for Lync client on Windows will be configured as a new Collaborated endpoint – the end result will be a Converged User with DN 70408.

In the case of a new Native User who will have no physical CS 1000 desk phone but will have a SIP endpoint on the Session Manager, a PCA configuration on CS 1000 is still required to direct the call from the Prime DN to the user's endpoint. So to add a new Native user, a PCA configuration must be added on CS 1000 for this purpose.

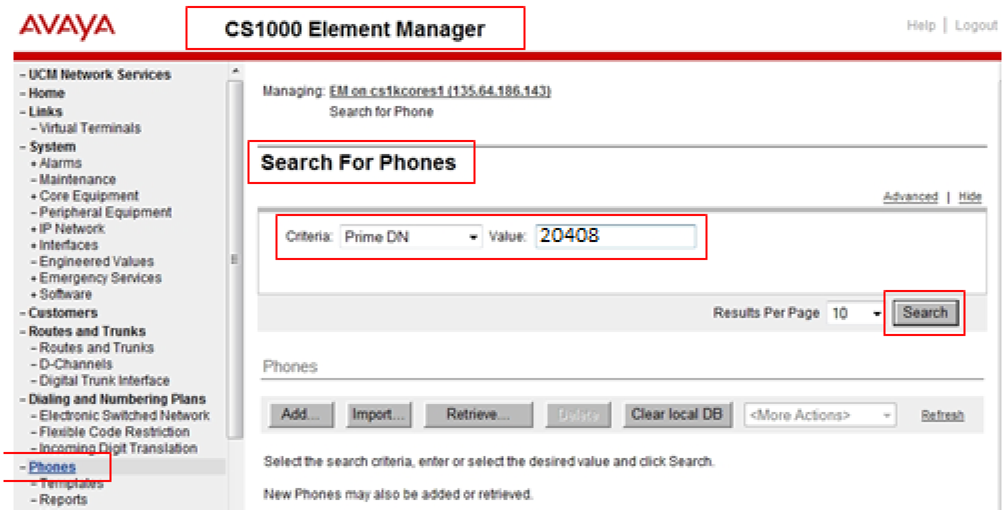
The following administration steps will be described:

- Confirm existing users in Avaya Communication Server 1000E Element Manager,
- Create User Identities and Communication profiles,
- Personal Call Assistant Configuration (PCA),
- Manual Configuration of Avaya SIP Clients.
- Synchronize CS 1000E Profile to User Identities in System Manager,
- Add PCA to CS 1000 User Communication profiles.

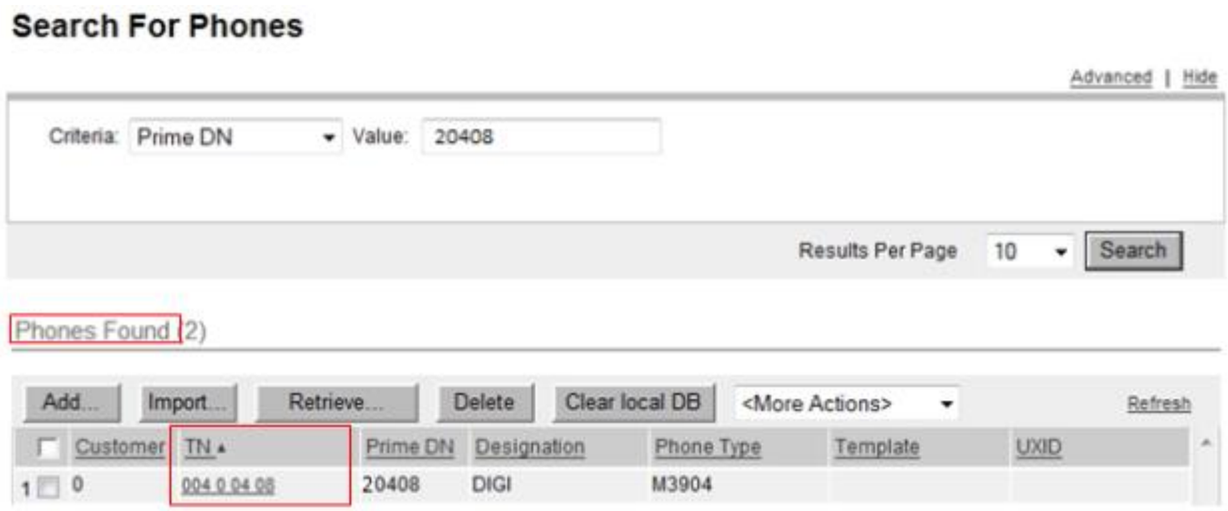
8.1. Confirm existing users in Avaya Communication Server 1000E Element Manager

It is presumed that existing CS 1000 users have been previously created using CS 1000 Element Manager which has configured the main endpoint phone Terminal Number (TN) / Directory Number (DN) and Call Party Name Display (CPND - First Name, Last Name). For each CS 1000 user defined in Element Manager a corresponding user identity must be added in System Manager. The **First Name** and **Last Name** of the user must match exactly on both CS 1000 Element Manager and System Manager User Management. This is important for proper Presence synchronization and import synchronization for user CS 1000 and CallPilot endpoint profiles.

Using the UCM Services link in the System Manager main page, access the **CS 1000 Element Manager** as described above in **Section 5.2**. Click on the link to **Phones**. In the **Search For Phones** window, the **Criteria** drop-down box is set by default to **Prime DN**. In the **Value** box, enter the prime DN of the existing CS 1000 user (e.g. **20408**). Click on **Search** button.



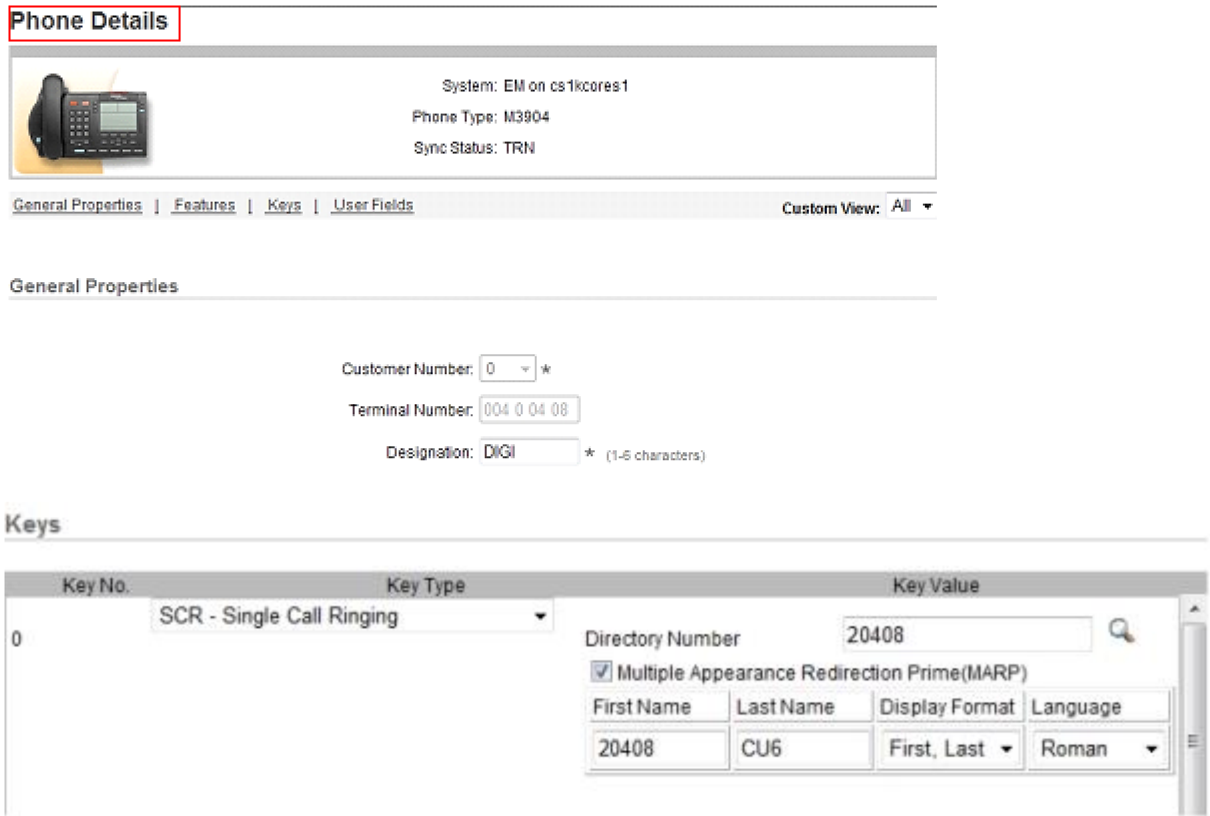
The result of the search comes back and displays **Phones Found**. Click on the TN hyperlink for the phone entry found (e.g. **004 0 04 08**).



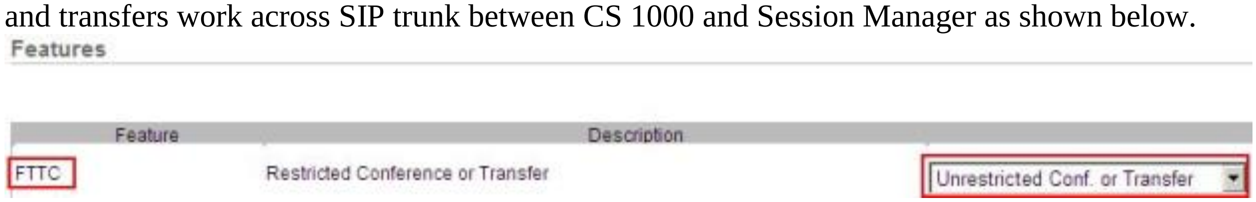
Search For Phones
Advanced | Hide
Criteria: Prime DN
Value: 20408
Results Per Page 10
Search
Phones Found (2)
Add...
Import...
Retrieve...
Delete
Clear local DB
<More Actions>
Refresh
Customer
TN
Prime DN
Designation
Phone Type
Template
UXID
1
0
004 0 04 08
20408
DIGI
M3904

(1)

The **Phone Details** page now appears. Scroll down to the **Keys** properties section and make note of the **First Name** and **Last Name** of the user (e.g. **20408**, **CU6**).

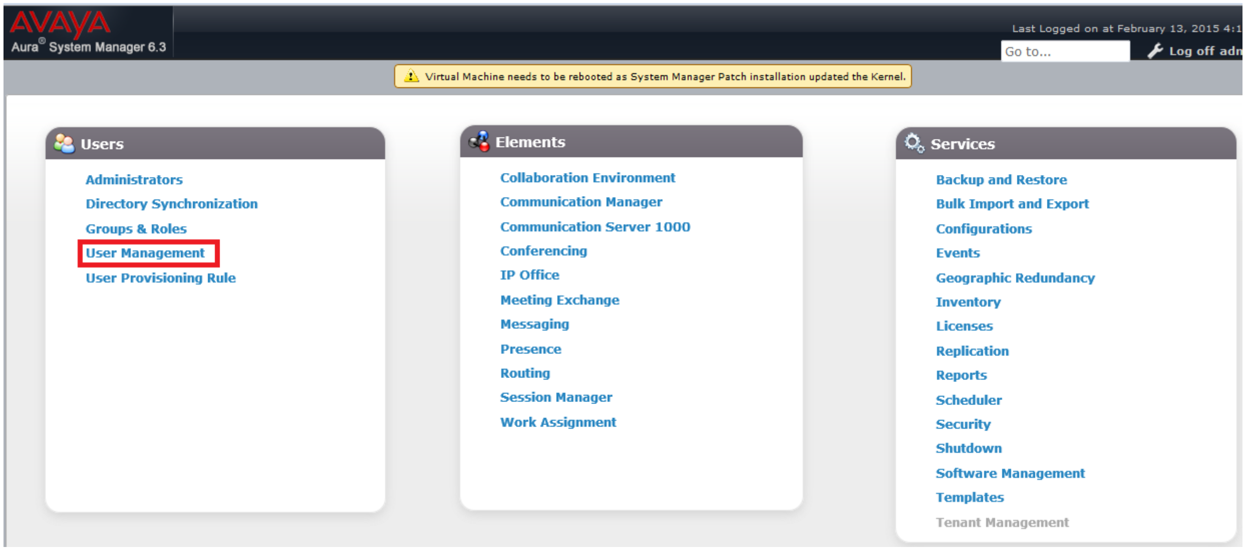


Ensure Feature FTTC is configured as "Unrestricted Conf. or Transfer" to ensure conferences and transfers work across SIP trunk between CS 1000 and Session Manager as shown below.

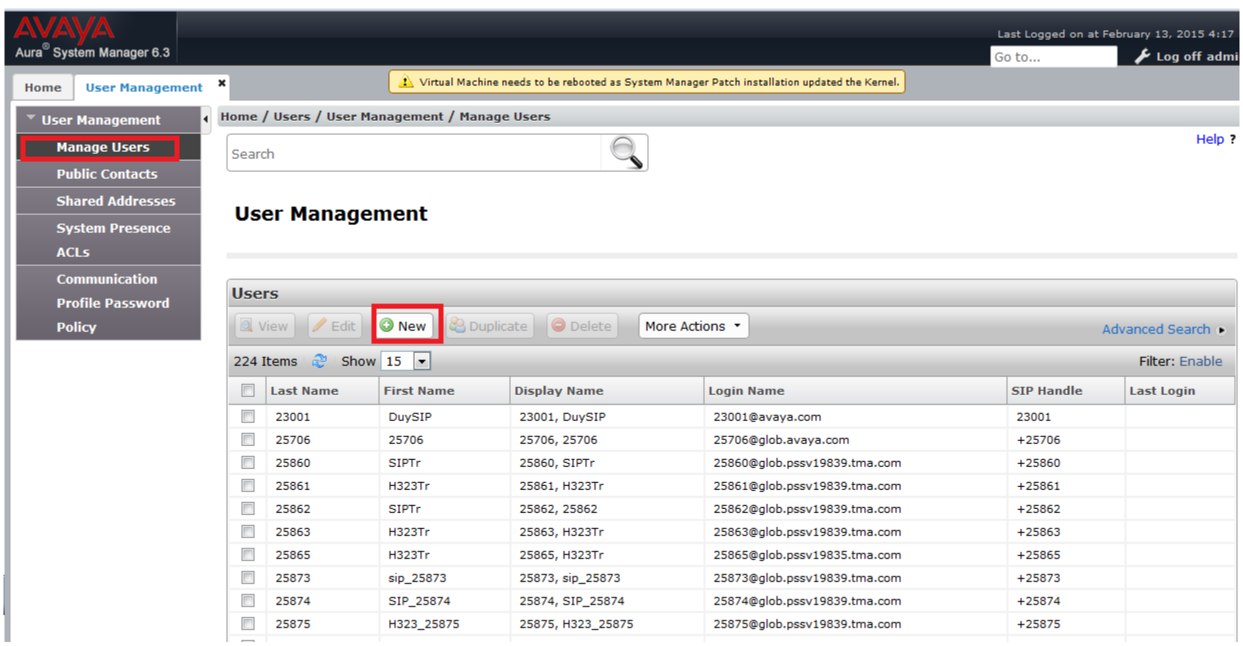


8.2. Create User Identities and Communication profiles

To create new users on System Manager, go to the main System Manager page and under **Users**, click on **User Management**.



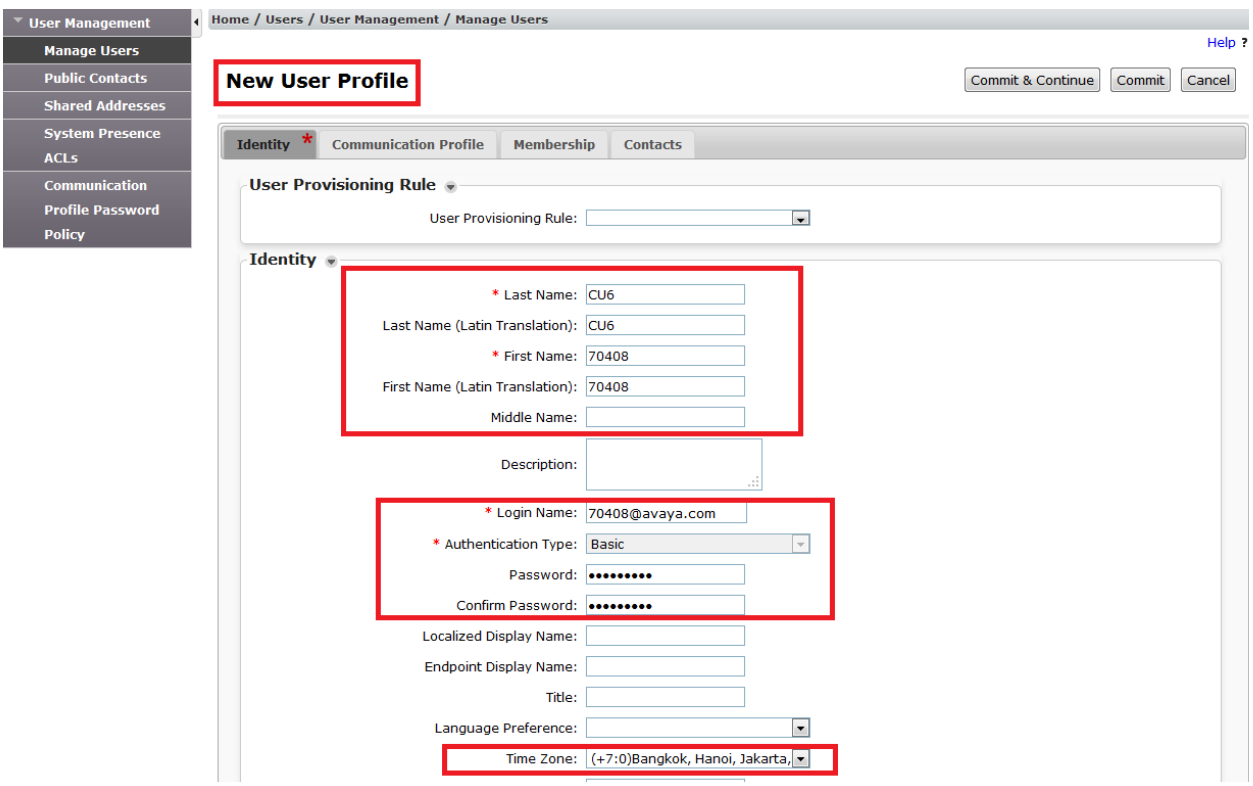
Under the **User Management** list on the left navigation menu, click on **Manage Users**. Click on **New** to add a new user.



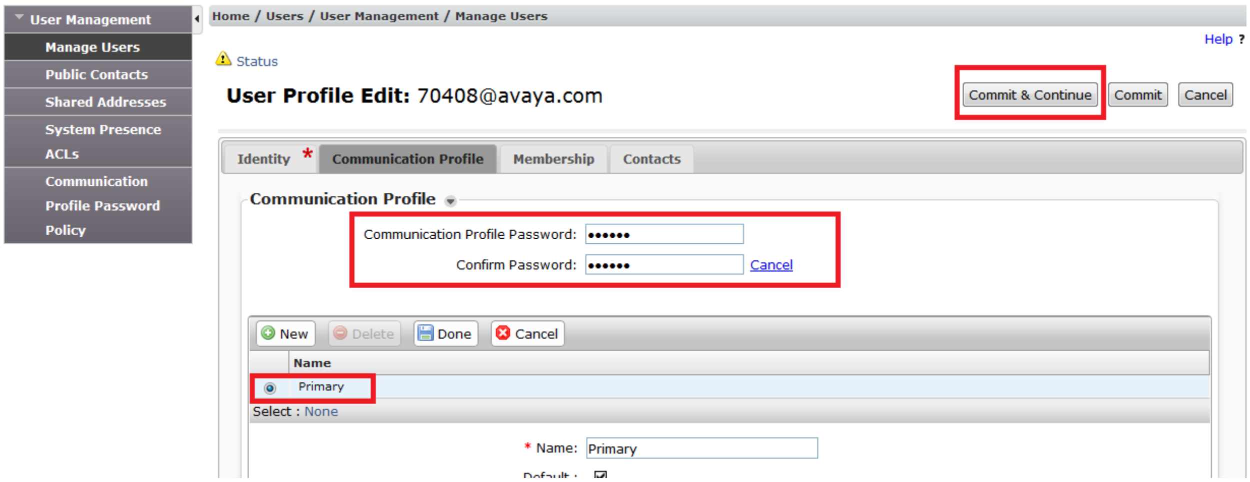
In the **New User Profile** page which appears, under the **Identity** section, enter values for the following required attributes for a new user and use default values for remaining fields.

- **Last Name:** Enter last name of user (e.g. **CU6** as noted above in **Section 7.1**).
- **First Name:** Enter first name of user (e.g. **70408**as noted above in **Section 7.1**).
- **Description:** Optionally enter a description.
- **Login Name:** Enter using the format "**handle@<domain>**" where "**<domain>**"matches the domain from **Section 5.1** (e.g. **70408@svstack.com)**.
- **Authentication Type:** Verify **Basic** is selected.
- **Password:** Enter the password used to log into System Manager.
- **Confirm Password:** Repeat password entered above.
- **Localized Display Name:** Enter a display name for the user (optional).
- **Language Preference:** Select the appropriate language from the drop-down list.

The field names marked with an asterisk (*) are mandatory fields. Before you click **Commit & Continue** ensure that all the mandatory fields have valid information.



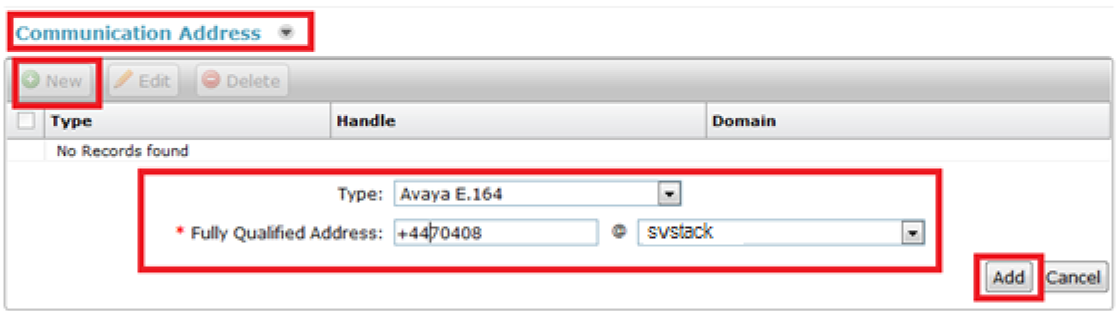
Next select the **Communication Profile** tab. Enter the password the user will use to register to Session Manager in the **Communication Profile Password** and **Confirm Password** fields (e.g. 123456 was used in the sample configuration). Verify there is a default entry identified as the **Primary** profile as shown below. Click on **Commit & Continue** to save this data.



Next**,** Expand the **Communication Address** sub-section and select **New** to define a **Communication Address** for the new user. Enter values for the following required attributes:

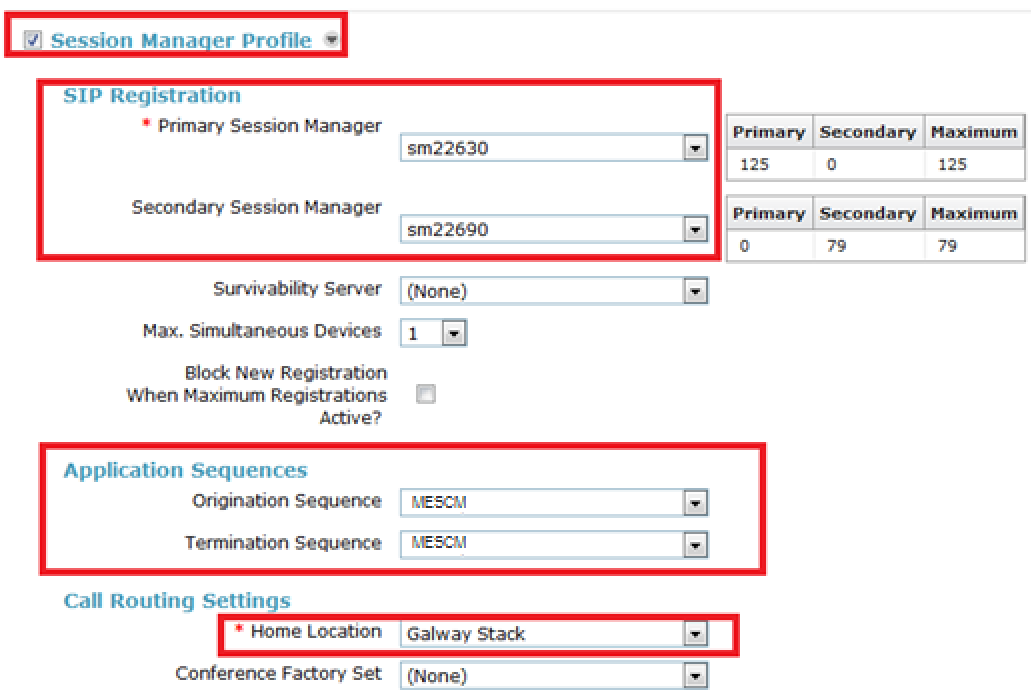
- **Type:** Select **Avaya E.164** from the drop-down menu.
- **Fully Qualified Address:** Enter extension number+4470408 **Domain:** Verify value matches Domain name defined in **Section 5.1** (e.g. **svstack.com**).

Click **Add** to save the Communication Address. **Note:** The Avaya XMPP communication address is added automatically to the communication profile because Presence Services is enabled as part of the Avaya Midsize Enterprise template. Telephony Presence will not display properly on a Collaboration client unless both XMPP and Avaya E.164 information have been entered.



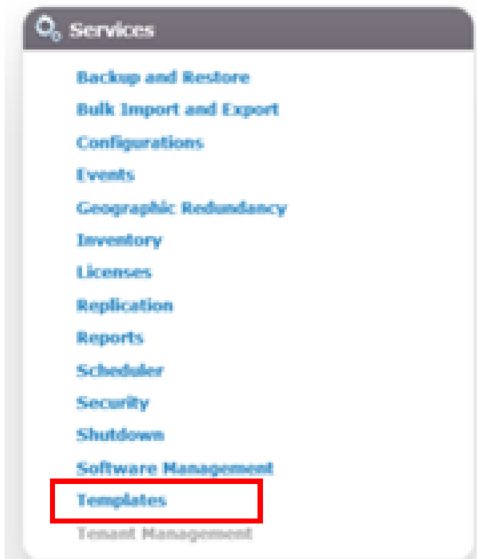
Scroll down to the **Session Manager Profile** section and expand this section. Enter the following required values and leave other values as default.

- **Primary Session Manager** Select a Session Manager from the drop-down list (e.g. **sm22630**).
- **Survivability Server** Select **(None)** from drop-down menu.
- **Origination Application Sequence** Select the Application Sequence defined for the Communication Manager from the drop-down list (e.g. **MESCM**).
- **Termination Application Sequence** Select the Application Sequence defined for the Communication Manager from the drop-down list (e.g. **MESCM**).
- **Home Location** Select a Location from the drop-down list (e.g. **Galway Stack**).
- **Conference Factory Set** Retain the default value of **(None)**.

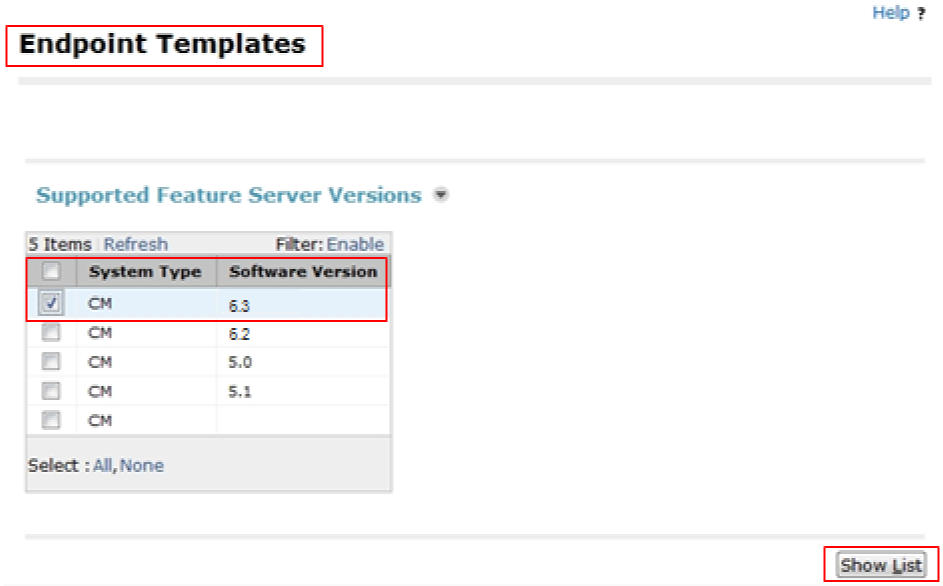


Scroll back up to the top of the page and click on Commit & Continue (not shown) to save this data before proceeding.

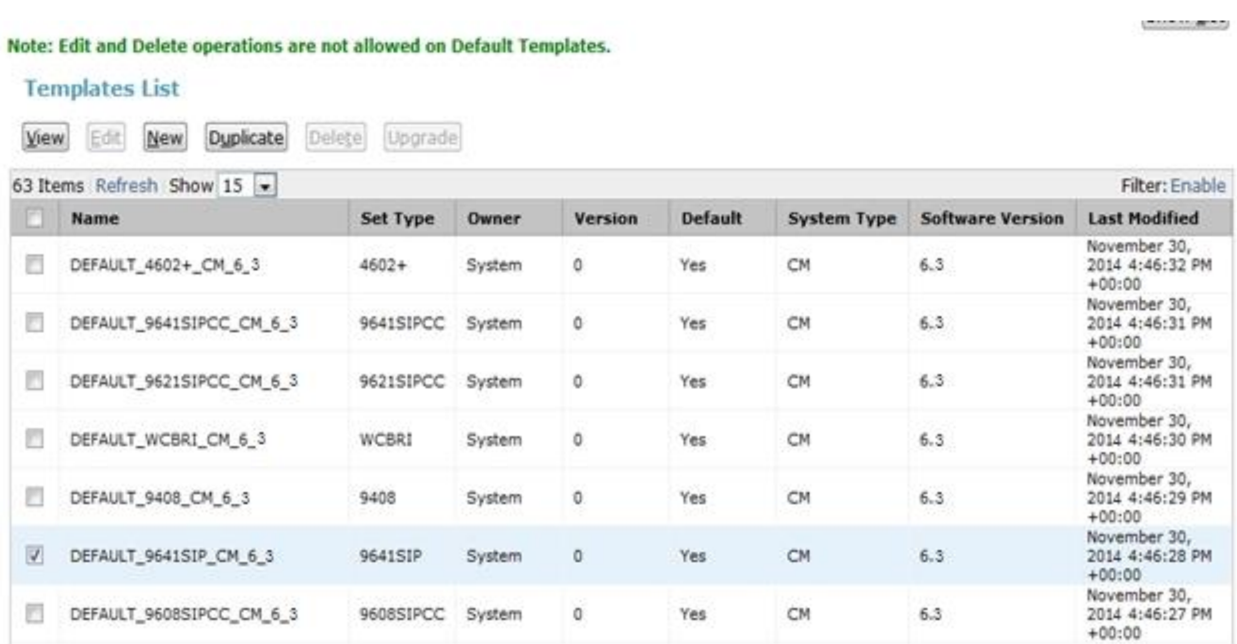
Prior to creating the Communication Manager endpoint profile for a non-standard user / endpoint (e.g. Avaya Communicator for Microsoft Lync), it is recommended to create a customized template. There is no standard template for Communicator for Lync type collaboration clients. Using the default 9641 SIP template, a duplicate may be made which can then be used for creating the Communication Manager endpoint profiles for these user types.
From the main System Manager page under the **Services** list, click on **Templates.**



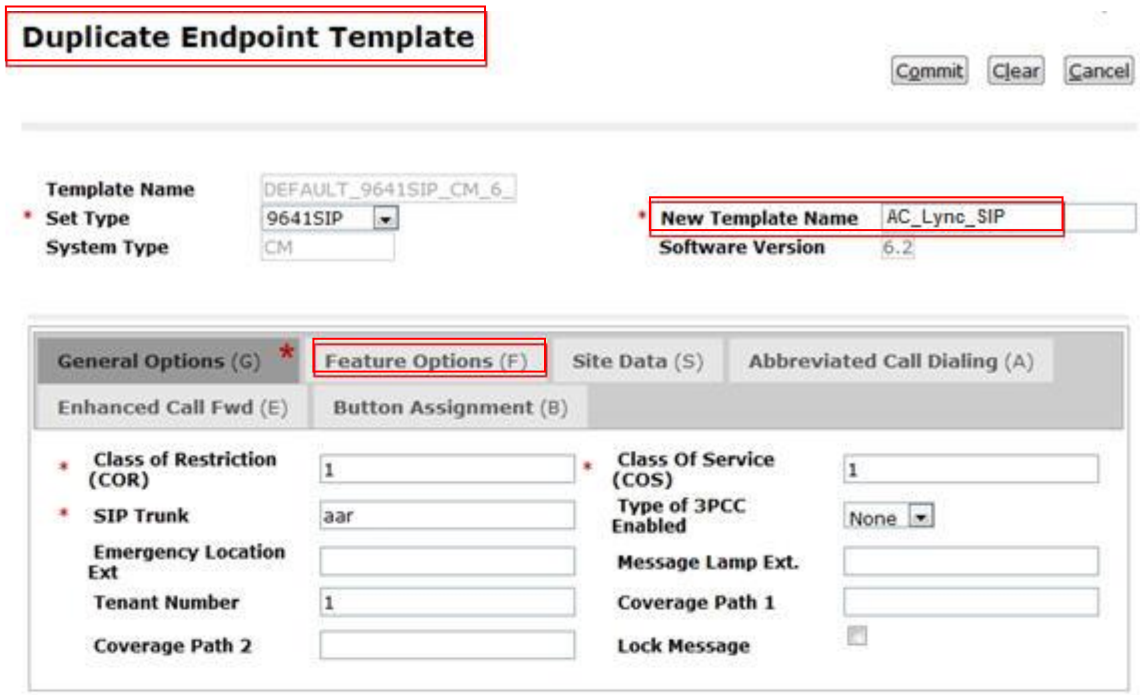
Under the **Templates** list on the left navigation menu, click on **CM Endpoint**. In the **Endpoint Templates** check the box for **System Type** and **Software Version** as **CM 6.2**. Click on **Show List**.



In the **Templates List** screen find the template called **DEFAULT_9641SIP_CM_6_3**, select the checkbox and the select the **Duplicate** button.

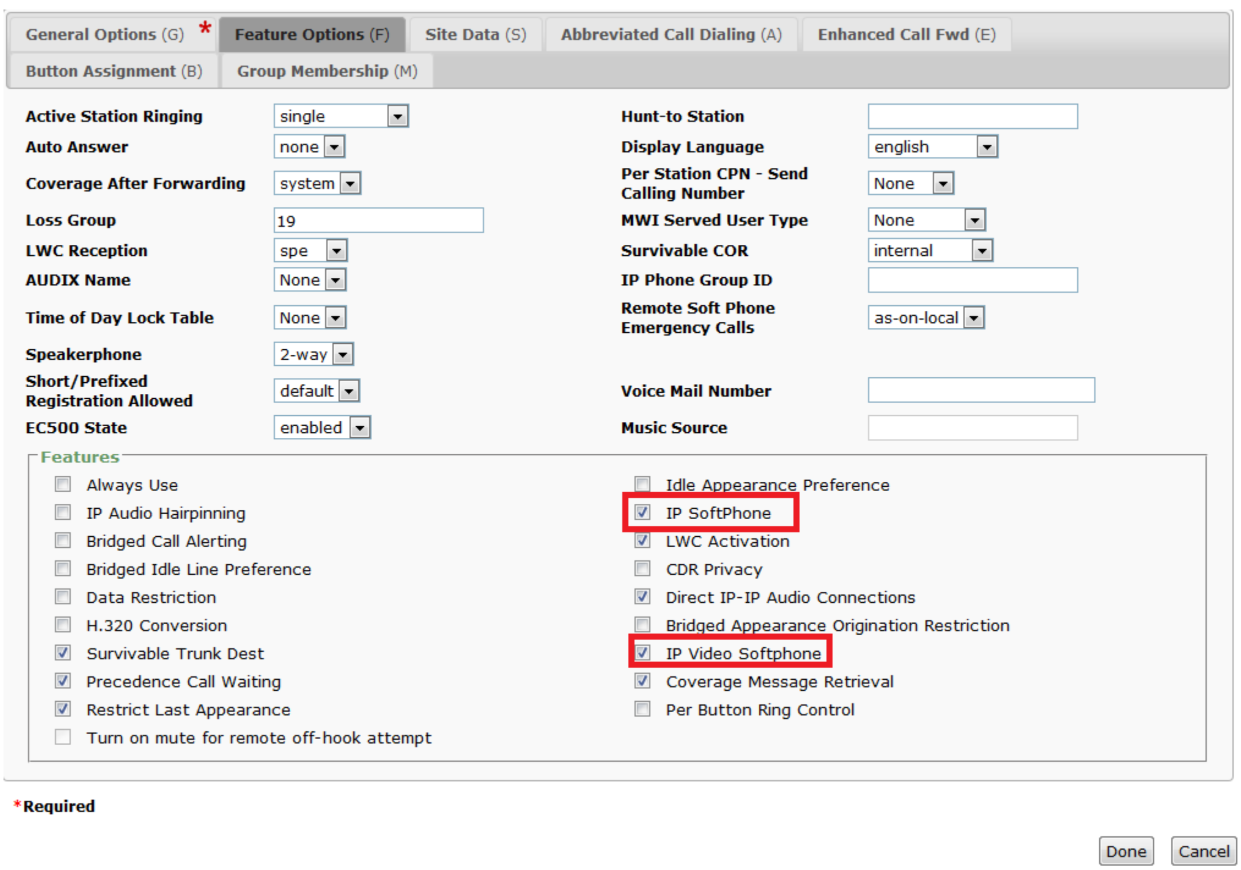


In the next screen, **Duplicate Endpoint Template**, enter a name in the **New Template Name** box (e.g. **AC_Lync_SIP**). Click on the **Feature Options (F)** tab.

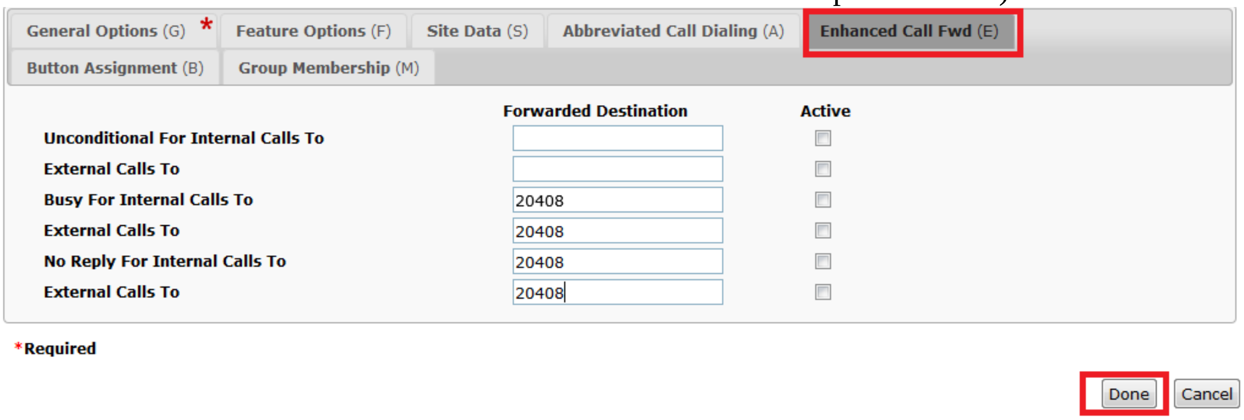


In the list of **Features**, tick the box for **IP Softphone and IP Video Softphone (if Video calling is required in Computer mode)**.

For Communicator for Lync user, CM extension should be enabled "call forward no answer" and



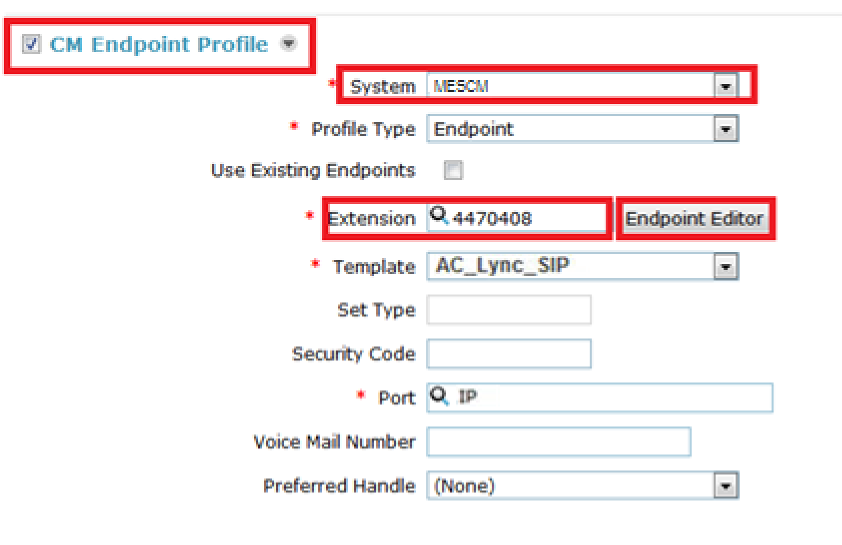
"call forward busy" to its CS 1000 desk phone (e.g. Communicator for Lync user 70804 has CM extension 4470408 and is set call forward to to its CS 1000 desk phone 20408)



Select the Commit & Continue button (not shown) to save the data.

Navigate back to User Management and Manage Users as described earlier in this **Section 8.2**. Edit the user "70408 CU6" and scroll down to the **CM Endpoint Profile** section and expand the section. Enter the following values and use defaults for remaining fields.

- **System** Select Managed Element defined for Communication Manager (e.g. **MESCM**).
- **Profile Type** Select **Endpoint**.
- **Use Existing Endpoints** Leave this box unchecked to automatically create a new endpoint when a new user is created.
- **Extension** Enter the CM extension number for the user (e.g. **4470408**).
- **Template** Select the template called **AC_Lync_SIP**
- **Security Code** Enter numeric value used to register the Communicator for Lync SIP endpoint. **Note:** this field should match the value entered for the Communication Profile Password above (e.g. 123456).
- **Port** Select **IP** from drop down menu.
- **Voice Mail Number** Leave this field blank.
- **Preferred Handle** Leave this set to (None).

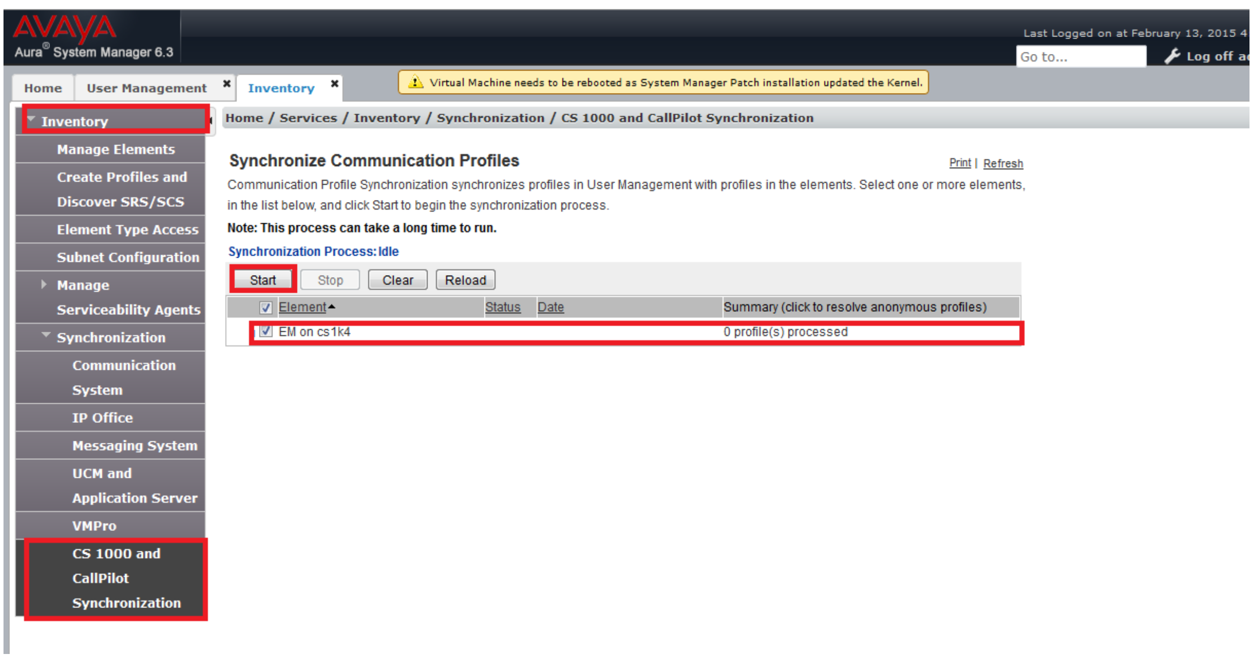


8.3. Synchronize Communication Profiles

System Manager provides an account synchronization feature to synchronize profiles between the different elements of the solution e.g. CS 1000, CallPilot, etc. It synchronizes profiles in User Management with the profiles in the respective elements. During synchronization, the account synchronization feature uses the account data in the elements as the master data. Therefore, when a profile data is not in synchronization with the element, the account data from the element is copied to System Manager.

8.3.1. Avaya Communication Server 1000E

Account synchronization with CS 1000 will import and synchronize all CS 1000 users into their previously created System Manager Identity CS 1000 Endpoint Profiles by matching each user's **First Name** and **Last Name**. After building the user identities, perform an on demand synchronization with CS 1000. From the System Manager home page under Elements click on Inventory (not shown). From the **Inventory** menu on the left hand side, under **Synchronization**, select **CS 1000 and CallPilot Synchronization**. Select a row associated with the CS 1000 and click on the **Start** button to initiate the synchronization process. Use the **Refresh** button in the table header to verify status of the synchronization. This synchronization process will add the CS 1000 Endpoint Profile to System Manager for each CS 1000 user name match.



8.4. Personal Call Assistant Configuration (PCA)

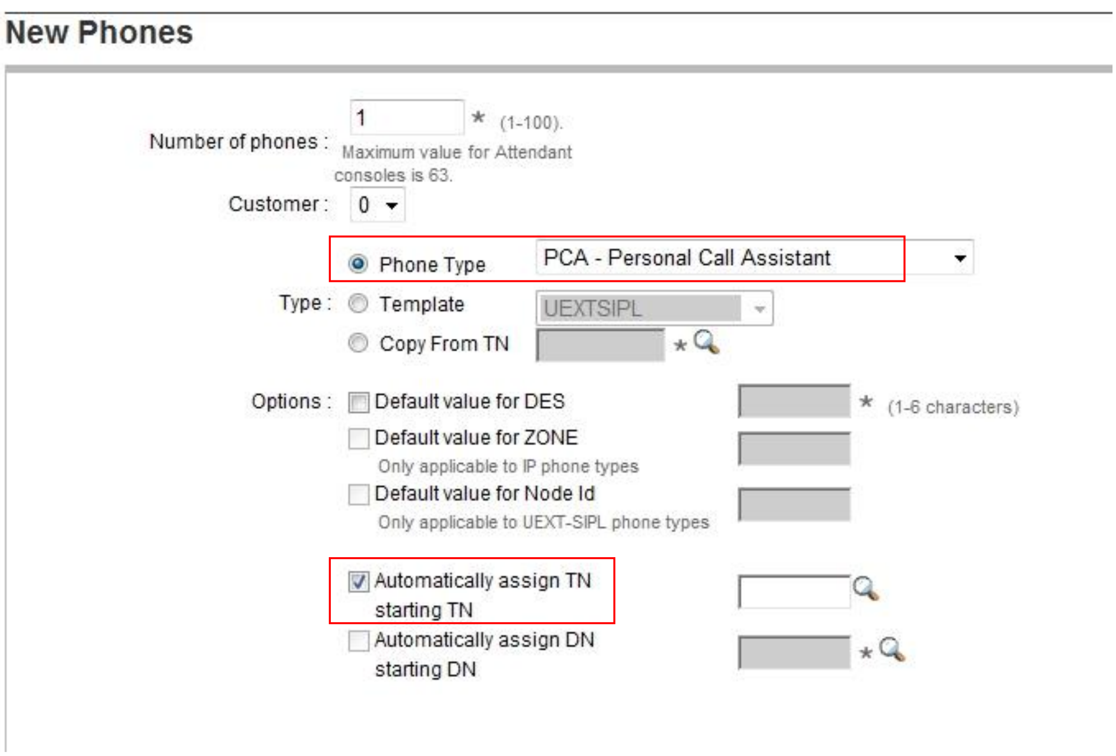
The PCA feature is utilized to enable the routing of calls for a CS 1000 user who is provisioned with Avaya Communicator Microsoft Lync integration on the Communication Manager. When calls are placed to the users published extension (DN: 70408) PCA will be used to route these calls to an Aura Extension 4470408. The steps below describe PCA configuration for an existing CS 1000 user (e.g. Prime DN 70408) to enable that user to be converged with a collaborated endpoint at 4470408.

To achieve this call routing a PCA must be configured per Communicator for Lync user. When there is a call on CS 1000to Communicator for Lync user, the PCA sends call signaling to the Avaya client endpoint via a CS 1000 SIP trunk to Session Manager. It is presumed that the PCA feature is licensed on the CS 1000 and enabled in the CS 1000 Customer Data Block. The following steps are required to configure a PCA:

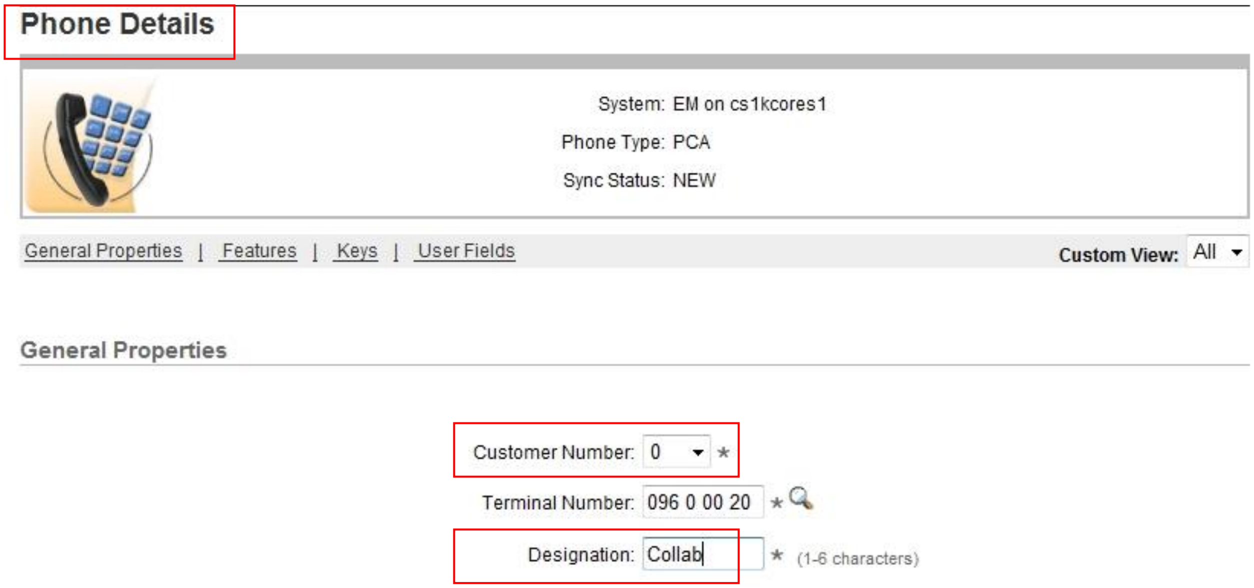
1. For each Communicator for Lync user:

- Configure key 0 as the Primary DN (e.g. 70408, note that CS 1000 desk phone DN for this user should be configured as 20408).
- Configure key 1 as HOT P key with the appropriate route prefix and DN as required to reach the twinned Avaya client endpoint. (e.g. 4470408)

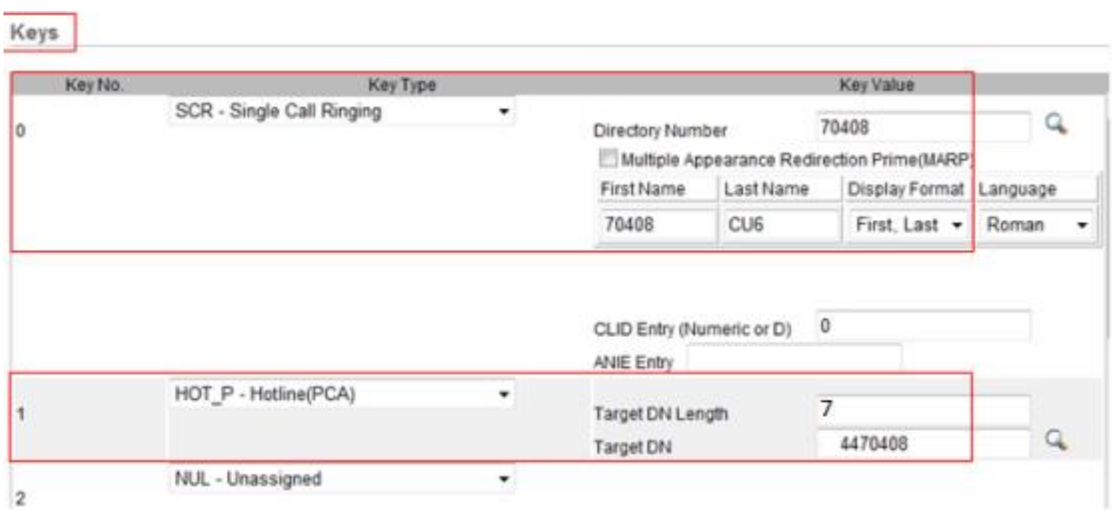
Step to add PCA: first navigate to “Phones” menu of CS 1000 EM page then click “Add”:
For **Phone Type** select **PCA-Personal Call Assistant** from the drop down menu. Next check the box to **Automatically assign TN starting TN**. Scroll down and select the Preview button (not shown).



The Phone Details screen appears. Enter a CS 1000 **Customer Number** (e.g. **0**) and **Designation** (e.g. **Collab**).



Scroll down to the **Keys** section. For **Key No. 0**, select **SCR – Single Call Ringing** from the drop-down **Key Type** list. In the **Directory Number** enter the prime DN for the user (e.g. **70408**). The **First Name** (e.g. **70408**) and **Last Name** (e.g. **CU6**) fields should automatically populate at this point. For **Key No. 1**, select **Hot_P – Hotline(PCA)** from the drop-down **Key Type** list. Enter the **Target DN Length** (e.g. **7**) and the **Target DN** of the Collaborated endpoint (e.g. **4470408**).



Select Commit (not shown) to save changes. When this is done, preform the account re-synchronization of the CS 1000 users to System Manager again using instructions in **Section 7.3.1**.

8.5. Configuring CLID for the User's CS 1000 desk phone

As described previously the Users CS 1000 desk phone has been configured with a new DN and this DN is used by Avaya Communicator for Microsoft Lync Other Phone Mode to make and receive calls on this device. While Communicator for Lync is controlling this desk phone the correct CLID for this user is presented to other users and externally.
If the user decides to make a call from the device then the CLID for this device i.e. 20408 will be exposed. To ensure the correct CLID is presented CS 1000 Load 15 is used to provide mapping of the existing DN to the correct CLID.

Example:
C1000 extn (PCA) = 70408
CM Extension of Lync = 4470408
CS 1000 Desk phone = 20408
E164 number +1 303 447 0408

In LD 15 Set create a new table enter this example is using 16

INTL Country Code = +1
Entry = 16
HNTN National Area Code = 303
HLCL Local Code for Listed Number = 447
DIDN_LEN DID Length = 4 (as the last 4 digits remain for the Lync extension and the CS 1000 phone and **0408** will be sent as part of the number)

```
>ld 15
CDB000
MEM AVAIL: (U/P): 94292520    USED U P: 8454965 904864    TOT: 103652349
SCH5066

REQ: chg
TYPE: net_data
CUST 0
OPT
AC2
FNP
CLID yes
  SIZE
  INTL 1
  ENTRY 16
    HNTN 303
    ESA_HLCL
    ESA_INHN
    ESA_APDN
    HLCL 447
    DIDN yes
    DIDN_LEN 4
    HLOC
    LSC
    CLASS_FMT
    ENTRY 16 SAVED!
  ENTRY
```

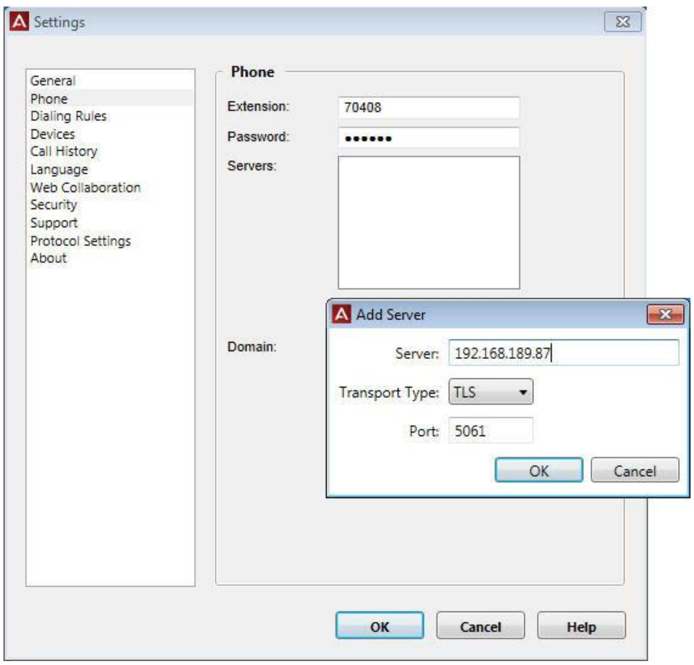
```
ld 20

PT0000
REQ: chg
TYPE: 1140
TN   96 0 0 13
ECHG yes
ITEM key 0 scr  20408  HNT FNA 16
        MARP
        CPND
        VMB
        ANIE
      KEY
ITEM
```

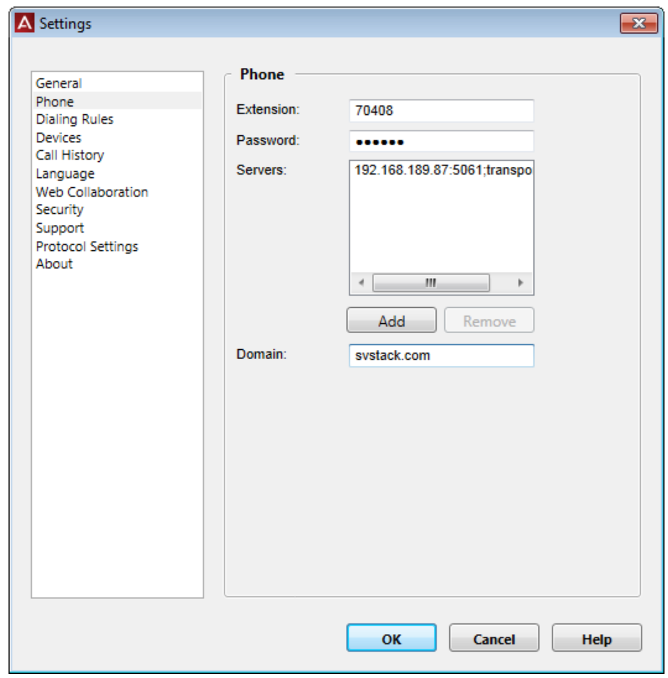
In LD 20 edit Key 0 of the phone to use this CLID table, configured as number 16 in this example from Ld 15 above. This desk phone DN is 20408

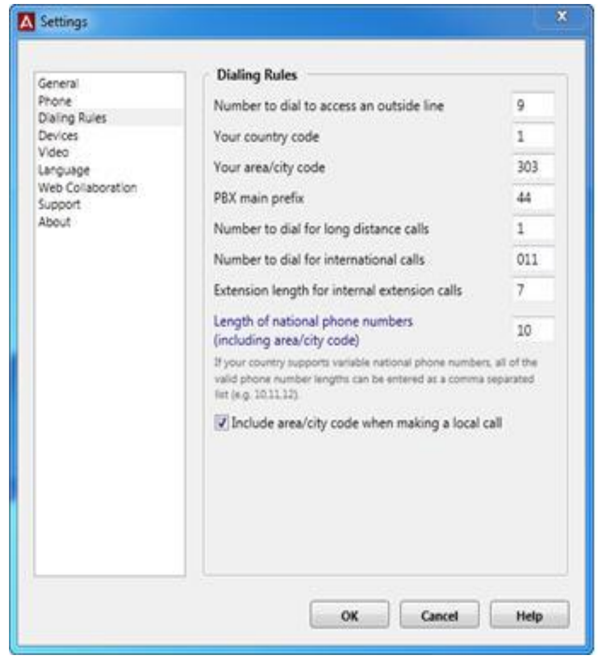
key 0 scr 20408 16

8.6. Manual Configuration of Avaya Communicator for Lync SIP client

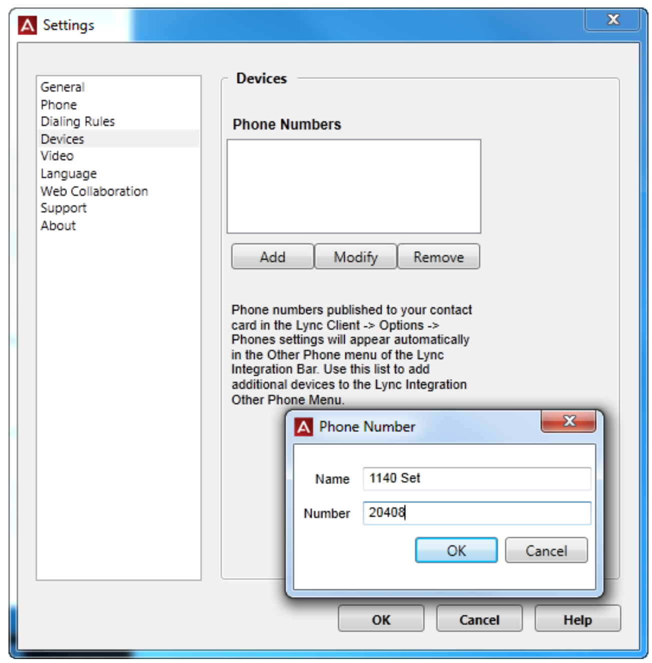
This section shows the steps required to manually configure a Communicator for Lync as an example of configuring a Collaboration endpoint / user to register to Session Manager.
It is presumed that the Communicator for Lync has been previously installed and the application has been launched. Click on the settings icon to open the **Settings** window. From the left hand list, select **Phone**. , Enter the Extension and Password and **Server address** of the Session Manager (e.g. **192.168.186.87, Port 5061, TLS)** and the domain name **svstack.com.** Click on OK to save the settings.



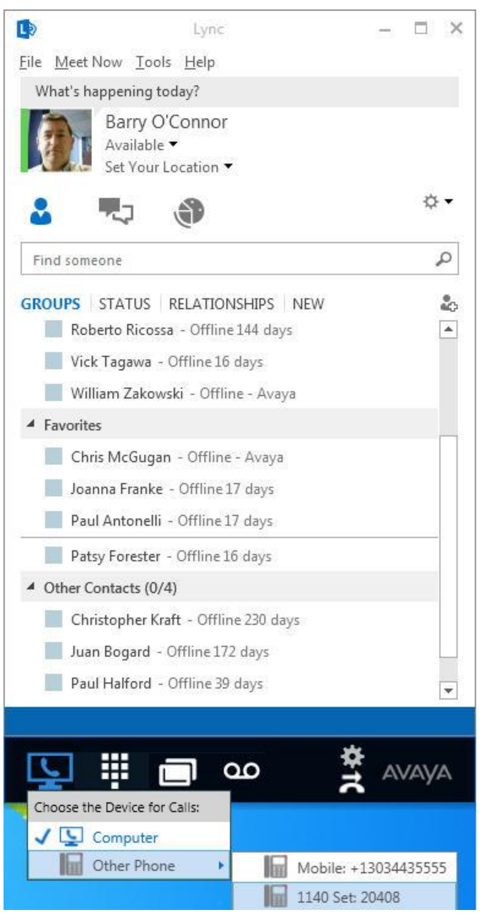




The following Dialing Rules are used for this configuration.



Under **Devices** select Add and enter a name and the number of the user's desk phone (20408).



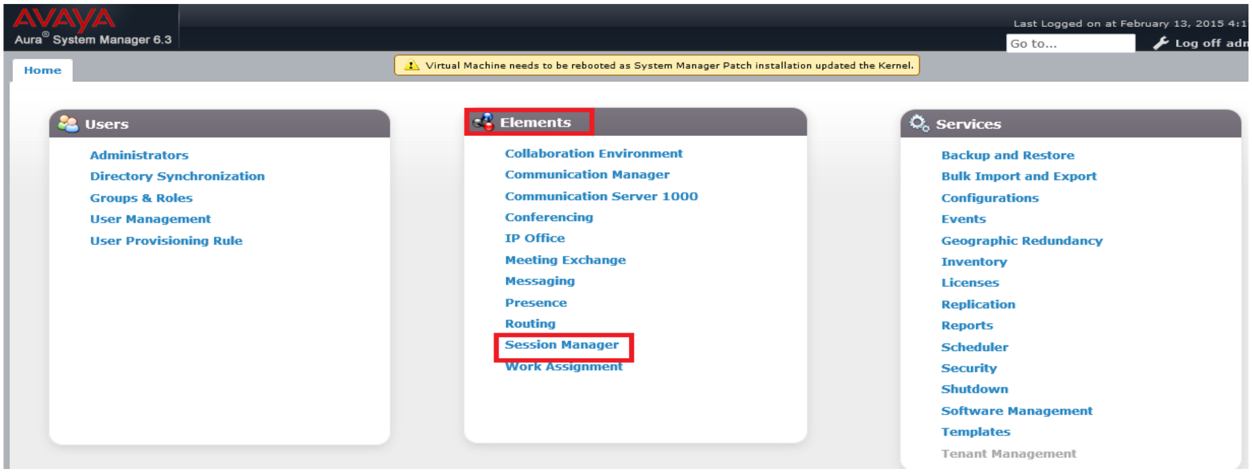
This device as well as other devices specified in Lync Options Phone such as their Mobile Phone will available to the user to select to make calls through under Other Phone Mode.

9. Verification Steps

To verify the status of some of the main elements in the solution, some checks can be carried out on Session Manager which is a core component in the integration of all the elements.

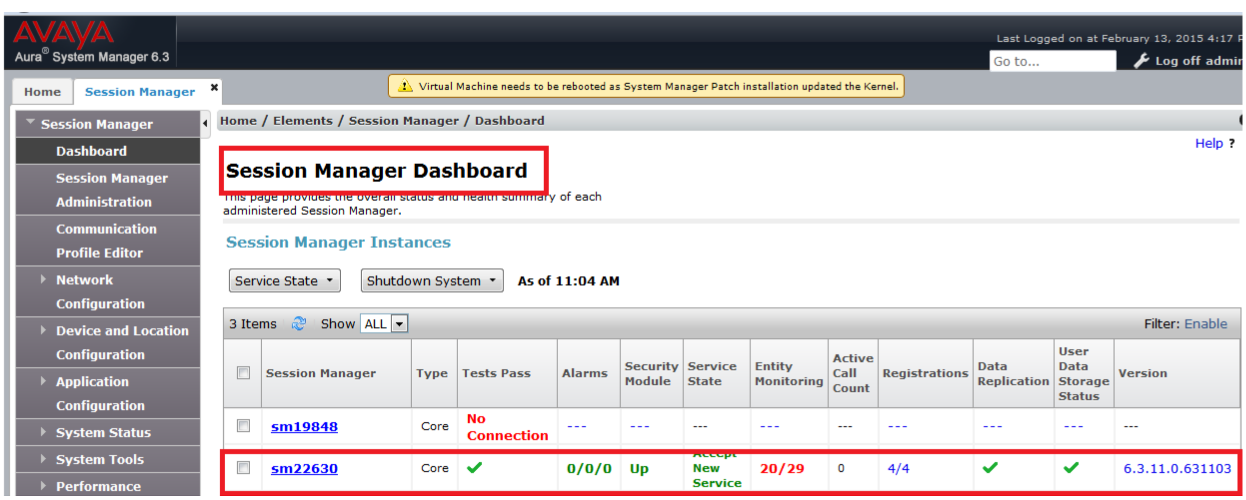
9.1. Verify Avaya Aura® Session Manager Operational Status

To verify Session Manager operational status, navigate from the main System Manager page under the **Elements** list, click on **Session Manager**.



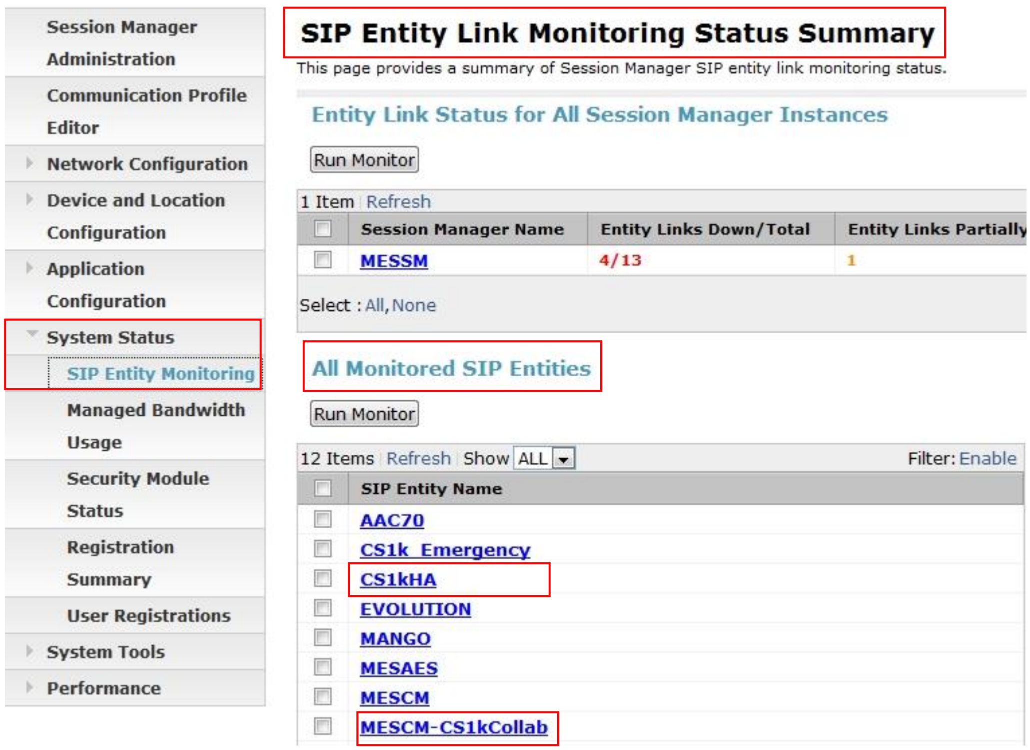
The **Session Manager Dashboard** window appears. For the Session Manager instance (e.g. **MESSM**), verify that the following fields:

- **Alarms** should show as 0/0/0 to indicate no alarms present,
- **Test Pass** should have a green tick-mark,
- **Security Module** should show as **Up**,
- **Service State** should show as **Accept New Service**,
- **Entity Monitoring** should ideally show a count indication of 0 entity down links / total links (in the example shown, there are 4 entity down links out of a total of 13 links – this is because this sample Session Manager has other entity links which are not up at this time).

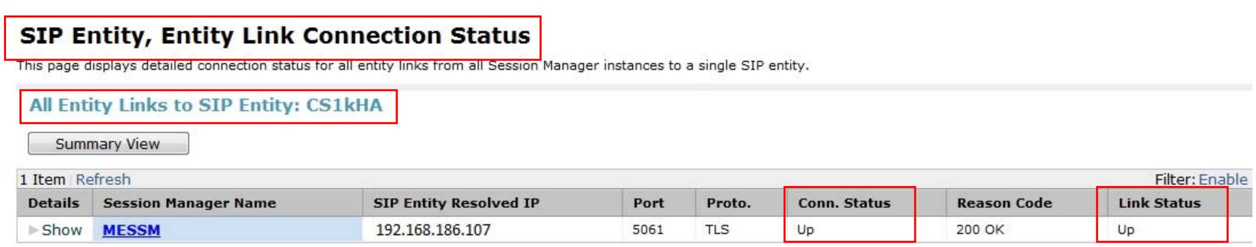


9.2. Verify Avaya Aura® Session Manager Entity Links Status

To further verify the Session Manager entity link status, click on **System Status** in the left hand list. Click on the link to **SIP Entity Monitoring**. The **SIP Entity Link Monitoring Status Summary** page appears. In the list of **All Monitored SIP Entities**, the sample configuration shows two entity links relevant to the CS 1000 / Aura solution. These are **CS1kHA** and **MESCM-CS1kCollab**. Click on the**CS1kHA** link first to check its status.



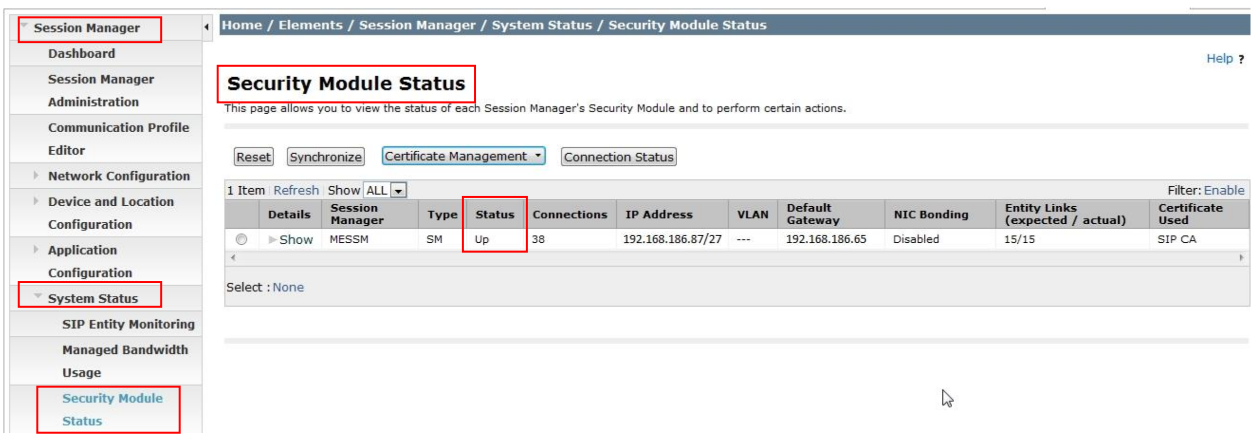
The **SIP Entity, Entity Link Connection Status** window appears. It shows **All Entity Links to SIP Entity: CS1kHA**. Verify that the **Conn. Status** is shown as **Up** and the **Link Status** is also **Up**.



Similarly, from the SIP Entity Link Monitoring Status Summary page, click on the MESCM-CS1kCollab link next to check its status (not shown). The **SIP Entity, Entity Link Connection Status** window appears. It shows **All Entity Links to SIP Entity: MESCM-CS1kCollab**. Verify that the **Conn. Status** is shown as **Up** and the **Link Status** is also **Up**.

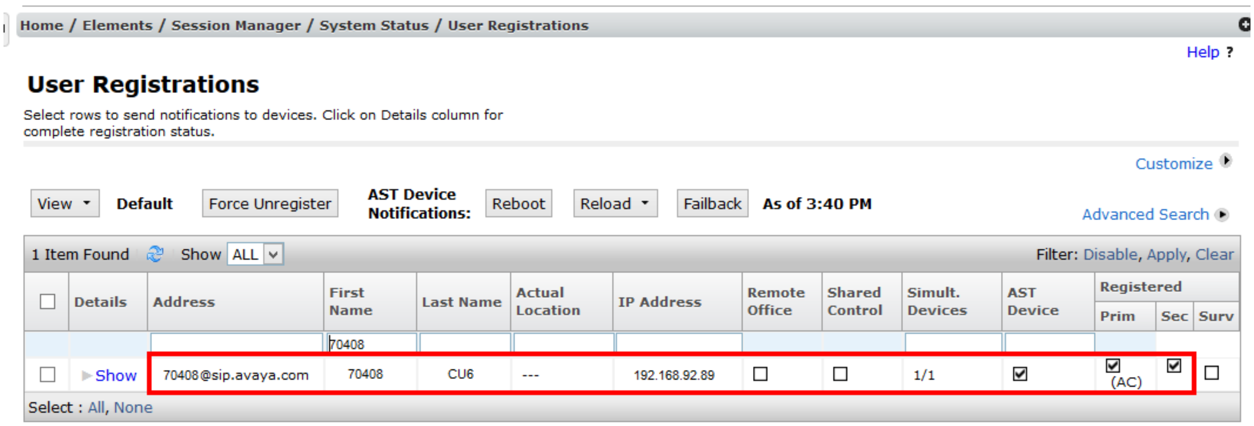
9.3. Verify Avaya Aura® Session Manager Security Module Status

Next verify the Session Manager Security Module status. From the **Session Manager** drop-down list on the left hand side, click on **System Status** and then on **Security Module Status**. In the **Security Module Status** window, verify the **Status** column displays **Up** as shown below.



9.4. Verify Registrations of SIP Endpoints

To verify that SIP endpoints have successfully registered with the Session Manager, perform the following check. From the **Session Manager** drop-down list on the left hand side, click on **System Status** and then on **User Registrations**. In the **User Registrations** window, verify the status of the sample endpoint which was successfully logged in per **Section 8.5** (i.e. user “**70408**, **CU6**”) by checking that it is registered with the primary (**Prim**) session manager.



10. Conclusion

This Application Note describes the configuration and test results for Communicator for Lync registered to Avaya Aura and interworking with CS 1000 Release 7.6.

The solution is made up of the either Avaya Aura® Midsize Enterprise system or discreet Aura component and Avaya Communication Server 1000E Release with CallPilot Release. This can be considered as a stepping stone for Avaya Communication Server 1000E installed base migration to a full Avaya Aura solution.

Within the test set up calls to and from Communicator for Lync users were performed with the following user types were successfully tested.

A list of the clients tested in the sample configuration is as follows:
CS 1000 Users:

- 11xx IP (UNISTIM) desk phone
- 12xx IP (UNISTIM) desk phone
- I2002p2/i2004p2 (UNISTIM) desk phone
- 39xx Digital desk phone

Avaya Communicator Microsoft Lync Users:

- Communicator for Lync clients registered to Communication Manager (SIP) making calls On Other Phone Mode through CS 1000 registered devices
    - 11xx IP (UNIStim) desk phone – CS 1000
    - 12xx IP (UNIStim) desk phone – CS 1000
    - I2002p2/i2004p2 (UNISTIM) desk phone – CS 1000
    - 39xx Digital desk phone – CS 1000

Aura Users:

- Avaya Communicator for Windows – ME
- Remote Avaya Communicator for Windows registered to ME via ASBCE
- One-X Communicator (SIP/SIP)

All testing was successful with the exception of those issues and limitations documented in **Section 1.2**.

11. Additional References

Relevant Application Notes:

A1. Application Note to administer voice mailboxes on Avaya CallPilot® R5.1 to provide shared messaging services for users in a CS 1000 Collaboration Pack solution.

A2. Configuring Secure SIP Connectivity using Transport Layer Security (TLS) between Avaya Aura® Communication Manager R6.3, Avaya Aura® Session Manager R6.3 and Avaya Communication Server 1000E R7.6.

A3. Application Notes for Configuring Converged and Native Users in a Collaboration Pack 1.1 for Avaya Communication Server 1000E Release 7.6

Additional Avaya product documentation is available at http://support.avaya.com.

Avaya Communicator for Microsoft Lync technical documentation

- Administrating Avaya Communicator for Microsoft Lync on Aura Release 6.4

Specific CS 1000Release 7.6 documentation guides relevant to this sample configuration are:

- Software Input Output Reference – Administration Avaya Communication Server 1000 (NN43001-611).
- Software Input Output Reference – Maintenance Avaya Communication Server 1000 (NN43001-711).
- IP Peer Networking Installation and Commissioning Avaya Communication Server 1000 (NN43001-313).
- Unified Communications Management Common Services Fundamentals Avaya Communication Server 1000 (NN43001-116).
- Element Manager System Reference – Administration Avaya Communication Server 1000 (NN43001-632).
- Emergency Services Access Fundamentals Avaya Communication Server 1000 (NN43001-613).
- Call Detail Recording Fundamentals Avaya Communication Server 1000 (NN43001-550).

Relevant Avaya Aura® documentations are also listed below for reference:

- ME Intelligent Workbook,
- Overview of Avaya Aura® Solution for Midsize Enterprise, Release 6.3,
- Implementing Avaya Aura® Solution for Midsize Enterprise Template Release 6.3,
- Installation and Upgrades for the Avaya G430 Branch Gateway,
- Administering Avaya Aura® Communication Manager(Doc ID 03-300509),
- Administering Avaya Aura® Communication Manager Server Options(Doc ID 03-
- 603479),
- Administrating Avaya Aura® System Manager,

A sample of Avaya CallPilot documentation guides relevant to this sample configuration are listed below for reference:

- Avaya CallPilot® Fundamentals Guide (NN44200-100)
- Avaya CallPilot® Library Listing (NN44200-117)
- Avaya CallPilot® Planning and Engineering Guide (NN44200-200)
- Avaya Meridian 1 and Avaya CallPilot® Server Configuration Guide (NN44200-302)
- Avaya Communication Server 1000 System and Avaya CallPilot® Server Configuration Guide (NN44200-312)
- Avaya CallPilot® Administrator Guide (NN44200-601)
- Avaya CallPilot® Software Administration and Maintenance Guide (NN44200-600)
- Avaya CallPilot® 202i Server Maintenance and Diagnostics Guide (NN44200-708)

A sample of documentation references relevant to the optional CS 1000 / Aura solution components of Avaya Aura® Conference and ASBCE are given below. These documents can be obtained from http://support.avaya.com.