

Avaya Solution & Interoperability Test Lab

Configuring Converged and Native Users in a Collaboration Pack 1.1 for Avaya Communication Server 1000E Release 7.6 - Issue 1.0

Abstract

This Application Note describes the detailed procedures for configuring Converged and Native Users with the "Collaboration Pack for Avaya Communication Server 1000". This solution consists of the Avaya Communication Server 1000 Release 7.6 and the Avaya Aura® Midsize Enterprise solution Release 6.2. In the sample configuration described herein, a Converged User is defined as a user having an Avaya Communication Server 1000E client and a Collaboration Pack client sharing a common Directory Number such that simultaneous ring (SIMRING), a single Avaya CallPilot voice mailbox and aggregated Presence are enabled for the user. A Native User is defined as a user having a Collaboration Pack client only, with no corresponding Avaya Communication Server 1000E client but calls are still routed via Avaya Communication Server 1000E. 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 and Interoperability Lab full stack testing and additional technical discussions.

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1. Introduction

This Application Note describes the procedures for configuring Collaboration Pack 1.1 for Avaya Communication Server 1000E Release 7.6 with Avaya Aura® Mid-Size Enterprise solution Release 6.2.

The "Collaboration Pack for Communication Server 1000" offer is based on Avaya Aura® Midsize Enterprise Release 6.2 and is used to accelerate the introduction of new collaboration clients to the Avaya Communication Server 1000 installed base. Collaboration clients tested in the sample configuration covered by this application note include Avaya SIP clients Flare® Experience for iPad, Flare® Experience for Windows, Avaya One-X® Mobile SIP for iOS, Avaya One-X Communicator (SIP) with Audio provided by a 96x1 SIP phone and 96x1 SIP One-X® Deskphone.

The notion of a Converged User is introduced with the Collaboration Pack for Avaya Communication Server 1000 (CS1000). A Converged User is defined as a user having both a traditional CS1000 client twinned with a collaboration client using the Personal Call Assistant (PCA) CS1000 feature. CS1000 clients tested in the sample configuration covered by this application note include the 11xx and 12xx UNIStim IP deskphones and 39xx digital deskphones. These two endpoints are seen by other users to be associated with a single user in that they share a common Directory Number (DN) for incoming and outgoing calls, they have a single voice mailbox on CallPilot and benefit from an aggregated presence functionality. A Native User is defined as a user having a Collaboration Pack client only with no corresponding CS1000 client but calls are still routed via CS1000 using the PCA feature. In contrast to the Converged and Native users, some existing users on the CS1000 may continue to operate without being twinned with a collaboration client. These are referred to as Non-converged Users. A Remote user is defined as a user/endpoint which is connected into the Collaboration Pack from outside the enterprise network using an optional Avaya SBC for Enterprise (ASBCE). The Remote user can therefore be either a converged or native user. In the case of a converged remote user, only the collaborated client is connected remotely via the ASBCE – the CS1000 client is not supported as a remote endpoint via the ASBCE.

Avaya Aura® Midsize Enterprise R6.2 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.

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® Midsize Enterprise R6.2 system is via SIP trunks and PSTN connectivity is provided through ISDN trunks on the CS1000.

Avaya CallPilot® R5.1 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 be enabled on CallPilot to allow

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the transmission of MWI to collaboration clients on the Avaya Aura® Midsize Enterprise system.

Dial-in and meet-me conference services only are provided with an optional Avaya Aura® Conference (AAC 7.0) server (i.e. Ad-hoc conference and dial-out services are not supported at this stage).

This Application Note will document the steps necessary to configure the main components of the Collaboration Pack solution to accommodate one new Converged User and one new Native 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 for Collaboration Pack,
2. Configure Avaya Aura® Session Manager,
3. Configure Avaya Aura® Communication Manager,
4. User Management – configure Converged and Native Users,
5. Verification Steps.

Detailed administration of other aspects of the CS1000 Collaboration Pack 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 and Avaya Aura® Presence Services for a Collaboration Pack solution are also not covered in this application note as are the administration of optional CS1000 Collaboration Pack solution components such as Avaya Aura® Conference (AAC), Avaya One-X® Client Enablement Services (CES), ASBCE for Remote User interactions, etc. References to relevant documentation sources are provided in **Section 11** to cover these.

Throughout this Application Note, the term “Collaboration Pack for Communication Server 1000” may sometimes be abbreviated and referred as “CS1000 Collaboration Pack”.

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2. Interoperability Testing

A reference configuration containing all of the equipment for the CS1000 Collaboration Pack was installed and a large number of tests cases were executed to ensure functionality of the various user endpoints supported and interoperability between CS1000 and Avaya Aura® Mid-Size Enterprise solution (i.e. the Collaboration Pack).

Converged user clients are "twinned" with one another via the Personal Call Assistant (PCA) feature on CS1000. The PCA feature or "PCA call" provides simultaneous ringing (SIMRING) of the two clients in the case of an incoming call to the CS1000 client. The user can answer on either client (CS1000 client or collaboration client on the Collaboration Pack) and the other will stop ringing. In the case of no answer or busy, call redirection will occur as defined for the CS1000 client which typically is a redirection to a single CallPilot voice mailbox. In the case of an outgoing call from the Converged User, in the case of the Avaya SIP client on the Collaboration Pack, the Calling Line Identification (CLID) feature of Communication Manager is used to send the CLID associated with the CS1000 client to ensure single number identity. For presence aggregation the telephony presence of the CS1000 client is pushed to the Avaya Aura® Presence Services application which also receives/pushes presence updates for the collaboration client

PSTN calling is achieved with ISDN trunks off the CS1000 for all clients.

It is presumed that the CS1000E Release 7.6 system and software and the CallPilot Release 5.1 integrated 202i server and software have already been provisioned. This combination represents an existing CS1000 Customer configuration. In addition, the extra components of the CS1000 Collaboration Pack solution such as Avaya Aura® Midsize Enterprise R6.2 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 Collaboration Pack solution (CS1000, Session Manager and Communication Manager) and subsequent user administration for one new Converged User and one new Native User.

2.1. Test Description and Coverage

To verify the interoperability and operation of Converged (and Native) Users the following features and functionality were covered during the testing:

- Simultaneous ringing for inbound calls (converged users only)
- Single 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
- Aggregation of presence status across devices for all user types
- Presence status during basic call scenarios
- Incoming and outgoing PSTN calls
- Point to point calls (call flows between each user / endpoint)

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- Multiple call scenarios
- User features such as hold and resume (with music on hold)
- Call transfer and conference calls
- Caller ID presentation during incoming and call transfers
- Proper codec negotiations (G711 / G729 / direct and in-direct media)
- Call Detail Recording (CDR) on the Communication Sever 1000
- CS1000 Attendant console interactions / call flows
- Basic emergency call functionality for all user types.

The following was not tested with the Collaboration Pack for CS1000:

- Any clients on the Collaboration Pack other than those explicitly referenced in this document are considered out of scope e.g. no H323 clients or no 96xx SIP clients
- Third party CDR applications
- AAC 7.0 multiple / cascading media servers
- AAC 7.0 Ad-hoc conference and Dial-out not supported for the solution at this stage
- No requirement to test SIP clients on the CS1000 as either Converged / Non Converged Users
- Support of a peer CS1000 system with the same ME (CS1000 Networking)
- Geo-redundancy and branch solutions
- PSTN calls routed directly via Collaboration Pack
- SIP Trunks to Service Provider
- Off-Net Call forwarding
- PRI or H323 trunks between CS1000 and Communication Manager for convergence
- FAX call testing
- Traffic / Load testing
- Scale testing with endpoints
- Internationalization / Localization testing
- Interactions with ACE
- Avaya Lync Integration
- Web Alive
- Identity Engine Analytics
- DTLS (secure signaling link) for UNIStim IP sets on CS1000
- Network Address Translation (NAT) remote endpoints through ASBCE
- Callpilot Desktop Messaging
- ACD agent as a Converged User
- CS1000 features which cannot be supported with Communication Manager, e.g. MADN (Multiple Appearance DN), Call Park, Call Pickup, Boss Secretary, etc

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2.2. Test Results and Observations

Interoperability testing of Converged and Native user endpoints for CS1000 Collaboration Pack solution was completed successfully with the following observations made and issues / limitations noted.

- Visual Voice Mail is not supported for one-X Mobile clients as it depends on Avaya Aura® Messaging, which is not part of the Collaboration Pack solution.
- Visual Conferencing is not supported for one-X Mobile clients as it depends on functionality which is available in AAC 6.0 but not yet available on AAC 7.0.
- Several issues were identified with One-X Mobile SIP for iOS and EC500 Simultaneous Ring feature. It is recommended to disable the EC500 Simultaneous Ring feature when One-X® Mobile SIP for iOS client is registered to Communication Manager.
- Significant known issues were found with Presence and Instant Messaging.
    - Telephony Presence for CS1000 non converged elements will time out from "Available" to "Unknown" after approx. 90 minutes if the user has not made a call. Once the user makes a call the Telephony Presence will again update to "On a Call" and subsequently to "Available".
- Direct Media needs to be disabled on Communication Manager for:
    - Music on hold to function properly,
    - Flare Experience clients (iPad & Windows) and Call Pilot.
- Need to disable Shuffling and Direct Media for remote users connected via ASBCE.
- Several display anomalies were seen in hold and transfer scenarios.
- Ad Hoc Conferencing via AAC7 & Dial Out from AAC7 is NOT supported.
- No failover capability for Flare® Experience and One-X Mobile SIP for iOS.
- Calls directed to converged users where the CS1000 phone (the prime phone in the converged pair) is busy will get Forward-Busy treatment and be sent to the appropriate CallPilot mailbox for that user. This treatment does not occur if the collaborated endpoint is busy while the prime CS1000 phone is idle – this is a known limitation of the PCA functionality on CS1000.

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3. Reference Configuration

The following diagram (**Figure 1**) shows the reference configuration used in the testing of a CS1000 Collaboration Pack sample solution. It depicts a possible CS1000 Collaboration Pack solution configuration.

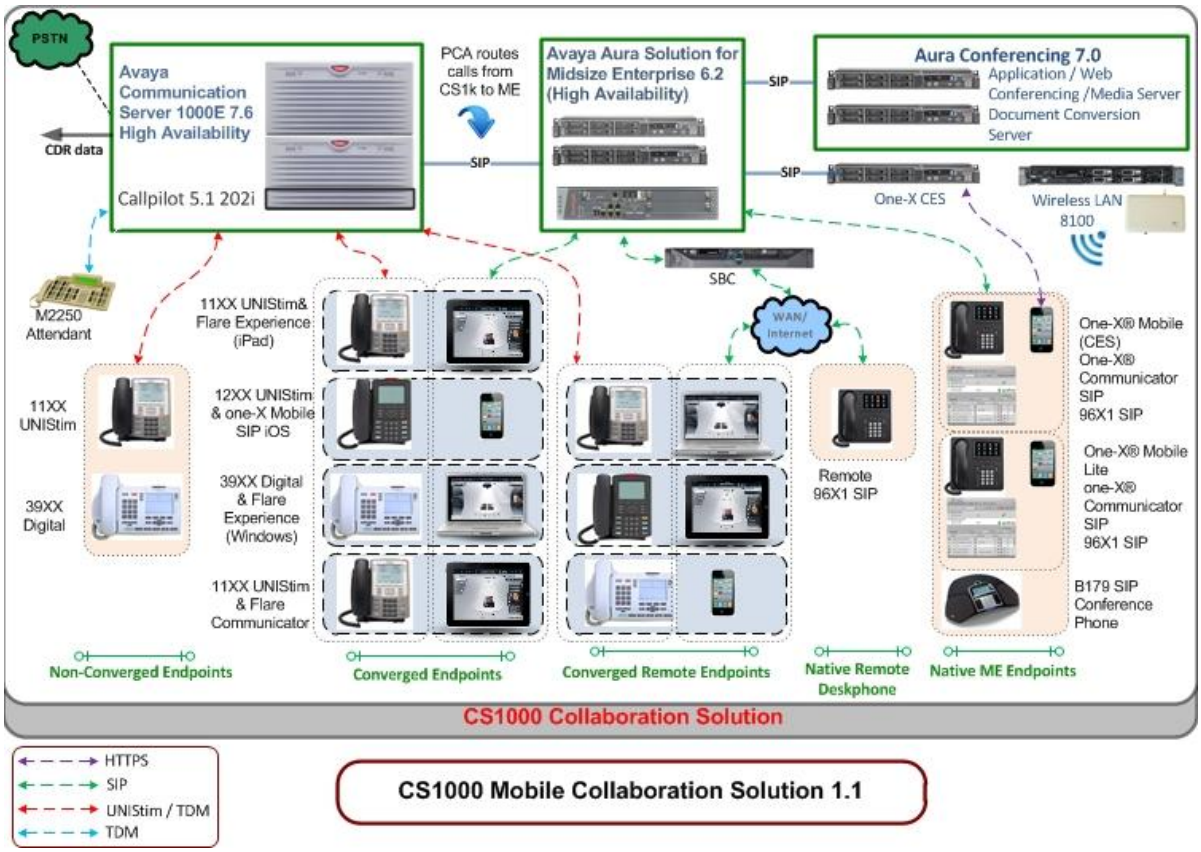


Figure 1: Typical CS1000 Collaboration Pack Solution Configuration

Three groups of users are defined as follows for CS1000 Collaboration Pack:

Group 1 - Non Converged User: An existing CS1000 user that continues to use their CS1000 desk phone with no association to any client on the ME / Collaboration Pack. A CallPilot voice mailbox is assumed to be already in place for a non-converged user. Supported endpoints are:

- 11xx IP (UNISTIM) deskphone
- 12xx IP (UNISTIM) deskphone
- 39xx Digital deskphone

Group 2 - Converged User: An existing CS1000 user that continues to use their CS1000 desk phone and is also associated, via PCA on the CS1000 side, with a SIP client on the ME Collaboration Pack. The endpoints in this arrangement are referred to as "Converged Endpoints". Some of these ME clients may be remote users who are connected into the Enterprise network

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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) deskphone (on CS1000)
- 12xx IP (UNIStim) deskphone (on CS1000)
- 39xx Digital deskphone (on CS1000)
- one-X Mobile (SIP) for iOS (on ME)
- Flare® Experience for iPad (on ME)
- Flare® Experience for Windows XP / 7 (on ME)
- Remote Flare® Experience for iPad registered to ME via ASBCE
- Remote Flare® Experience for Windows registered to ME via ASBCE
- Remote one-X® Mobile SIP for iOS registered to ME via ASBCE

Group 3 - Native User: A user on the ME Collaboration Pack that has no corresponding CS1000 desk phone. Services for these users are provided via clients on the Collaboration Pack only. Incoming PSTN calls or calls from non-converged users are routed to the ME over SIP trunk from CS1000 via PCA. The endpoints in this arrangement are referred to as "Native Endpoints". Some of these ME 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® Deskphone 96x1 SIP
- Avaya One-X Communicator (SIP) with Audio provided by an Avaya one-X® Deskphone 96x1 SIP phone and 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® Deskphone 96x1 SIP on ME only (i.e. not supported for a CS1000 phone).
- Avaya one-X® Deskphone 96x1 SIP
- Remote Avaya one-X® Deskphone SIP on 96x1 registered to ME via ASBCE

Other components of the solution are also shown in **Figure 1** above such as AAC 7.0, One-X CES and Wireless LAN 8100 infrastructure and a description of how these are configured for CS1000 Collaboration Pack is outside of the scope of this application note. **Section 11** has references to other documentation guides and application notes relevant to the CS1000 Collaboration Pack solution covering topics such as configuration of CallPilot voicemail for converged and native users in a CS1000 Collaboration Pack solution.

This Application Note describes the configuration of the various system components required to enable an existing non-converged CS1000 user to be converged (twinned) with a collaborated client. It also covers the configuration required for provisioning native users on the ME system such that calls from anywhere in the system are routed through the CS1000 via PCA to the ME system. A requirement of the CS1000 Collaboration Pack solution is that calls from either of the converged endpoints (i.e. CS1000 client or the twinned collaborated SIP endpoint) must be presented as a single number / Calling Line ID (CLID). So even though the collaborated endpoint may have a different number on the ME system than its twin on the CS1000, the CLID

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/ number presented must be the one from the CS1000. In the case of a native user where there is no physical endpoint on the CS1000 system, a user profile must exist on the CS1000 to reflect the correct phone number / CLID. This profile on CS1000 also allows call redirection to CallPilot for voicemail services.

For the purposes of this application note, a more simplified diagram of the CS1000 Collaboration Pack 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 converged user in the sample configuration has been designed such that a DN for any endpoint on the Collaboration Pack side is the same as the DN of its "twin" on the CS1000 side plus a prefix of 44. So in this example, a 3904 TDM telephone set has been provisioned on the CS1000 system with a DN = 70408 and a Flare Experience on Windows endpoint on the Avaya Aura® Midsize Enterprise system has been provisioned with a DN = 4470408. The number routing and dial plan manipulations to allow the twinning of these two endpoints are handled by the CS1000 and Avaya Aura® Midsize Enterprise systems. The overall result is that any call made to converged user 70408 will automatically get routed to both of these endpoints.

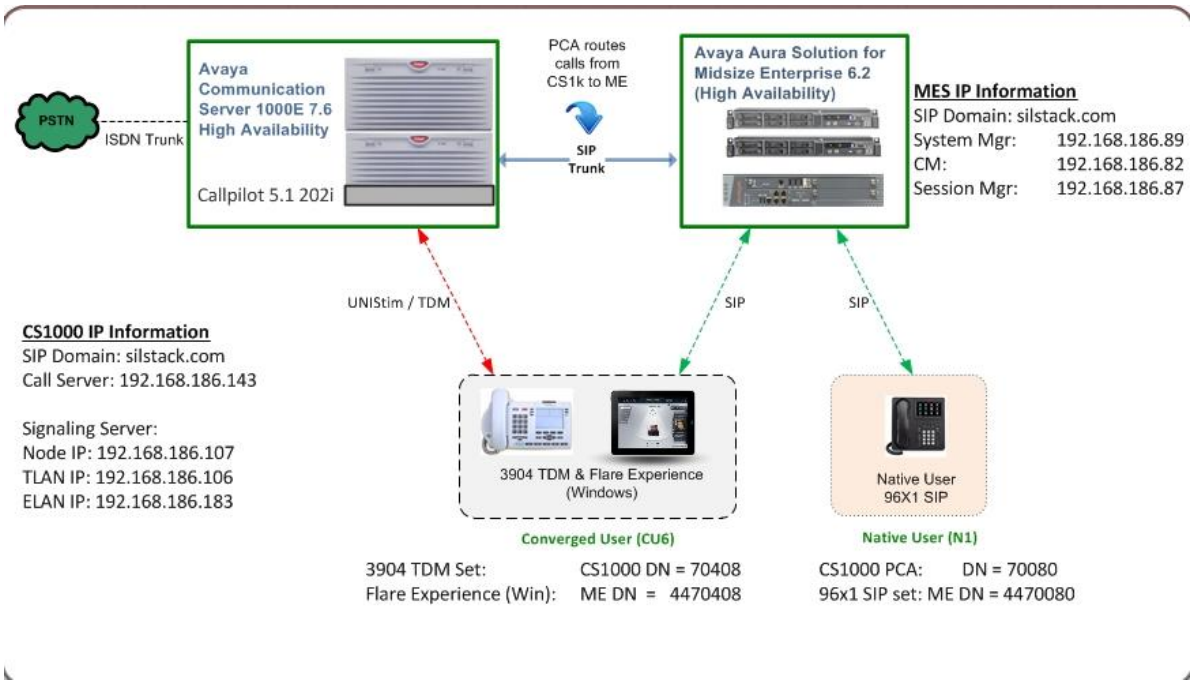


Figure 2: Simplified CS1000 Collaboration Configuration

Similarly, for a native user who has a SIP set on the Avaya Aura® Midsize Enterprise system e.g. a 96x1 SIP set with DN 4470080, there is a PCA configuration on CS1000 which has a DN of 70080. As all calls are routed through the CS1000, then any call to 70080 will automatically get routed via the PCA to the 96X1 SIP set.

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4. Equipment and Software Validated

The following equipment and software were used for the sample CS1000 Collaboration Pack solution test configuration.

Equipment/Software	Release/Version
Avaya Communications Server 1000E PBX	R7.6 + latest patches.
Avaya CallPilot running on a 202i integrated server	R5.01.01 + PEPs CP0501SU001S, CP501S01G08S, CP501S01G09C
Avaya 1100 Series IP Telephones for Avaya Communications Server 1000E	Firmware version 5.5 (UNIStim)
Avaya 1200 Series IP Telephones for Avaya Communications Server 1000E	Firmware version 5.5 (UNIStim)
Avaya 3900 Series TDM Telephones for Avaya Communications Server 1000E	Firmware version AA94 delivered with CS1000 R7.6
Avaya Aura® Solution for Midsize Enterprise (ME) running on an Avaya S8800 ME server platform	Avaya Aura® Communication Manager R6.2 SP4 Avaya Aura® System Manager R6.2 SP4 Avaya Aura® Session Manager R6.2 SP4 Avaya Aura® Presence Services R6.1 SP5
Avaya SBC for Enterprise running on a Dell R200 server	R6.2.0.Q33
Avaya Flare® Experience for iPad running on Apple iPad (OS 6.0)	R1.1
Avaya Flare® Experience for Windows XP running on an HP PC	R1.1
Avaya One-X® Mobile SIP for iOS running on an Apple iPhone4/4s	R6.2
Avaya one-X® Mobile with Client Enablement Services (CES)	R6.1
Avaya one-X® Mobile Lite	Rel. 1.5.5
Avaya one-X® Deskphone 96x1 SIP with One-X® Communicator	96x1 Rel. 6.2 One-X Communicator Rel. 6.1.7
Avaya one-X® Deskphone 96x1 SIP	R6.2
Avaya B179 SIP Conference Phone	R2.3.1

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5. Configure Avaya Communication Server 1000E

This section describes the details for configuring CS1000E) to route calls to the Collaboration Pack as well as provide simultaneous ringing to Avaya Aura® SIP endpoints via Session Manager over a SIP trunk. These instructions assume that the CS1000E has been registered as a member of the System Manager Security framework. In addition, these instructions also assume that the configuration of the CS1000E Call Server and Signaling Server applications has been completed to support SIP trunks, IP (UNIStim) telephones and Digital telephones. Refer to **Section 11** for more information on how to administer these functions.

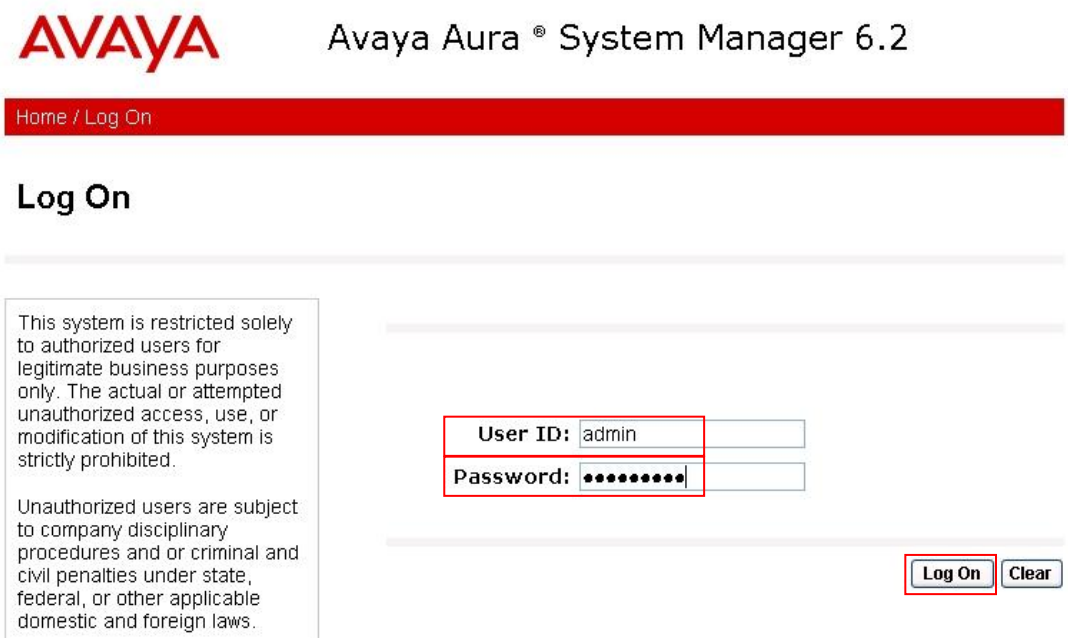
Using the Avaya Unified Communications Management (UCM) interface, the following administration steps will be described:

- 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

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5.1. Logon to Avaya Aura® System Manager

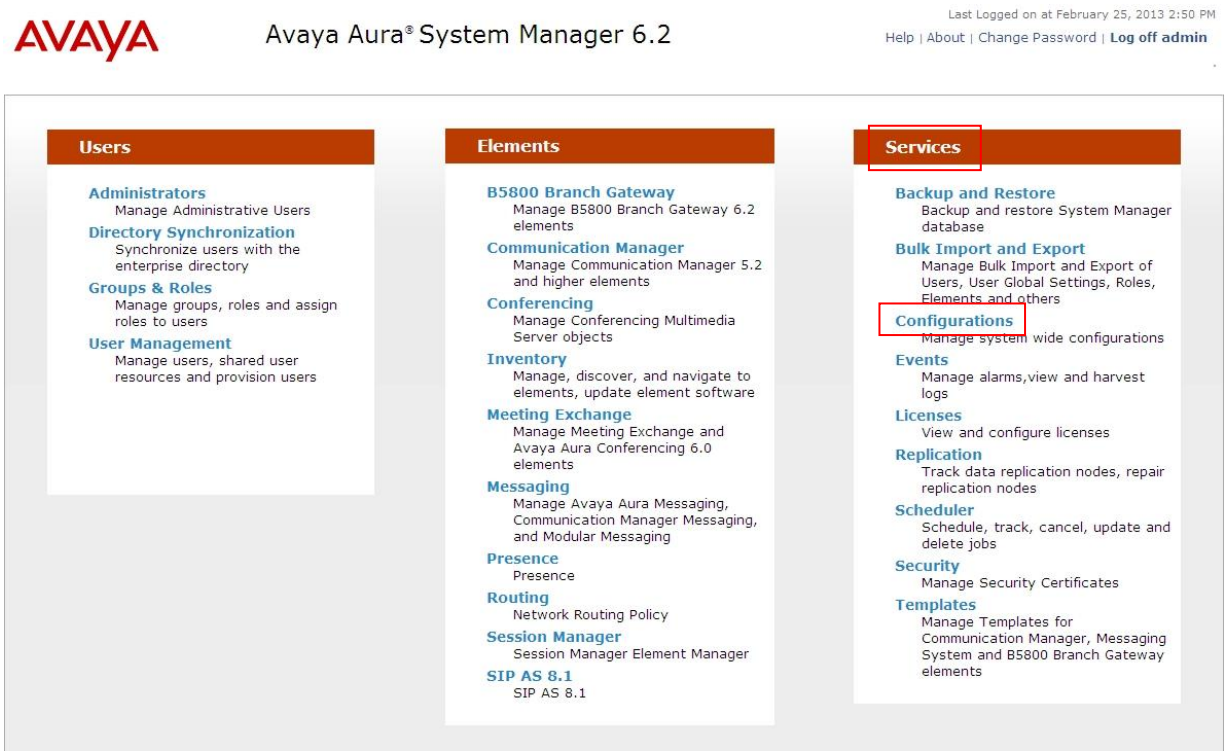
Using any supported browser, access the web based GUI of System Manager by using the URL **https://<FQDN>/SMGR**, where **<FQDN>** is the Fully Qualified Domain Name of System Manager (for example https://messmgr.silstack.com/SMGR). In the **User ID** box, enter **admin** and enter the corresponding admin account password in the **Password** box. Click on the **Log On** button to login to the System Manager 6.2 console.



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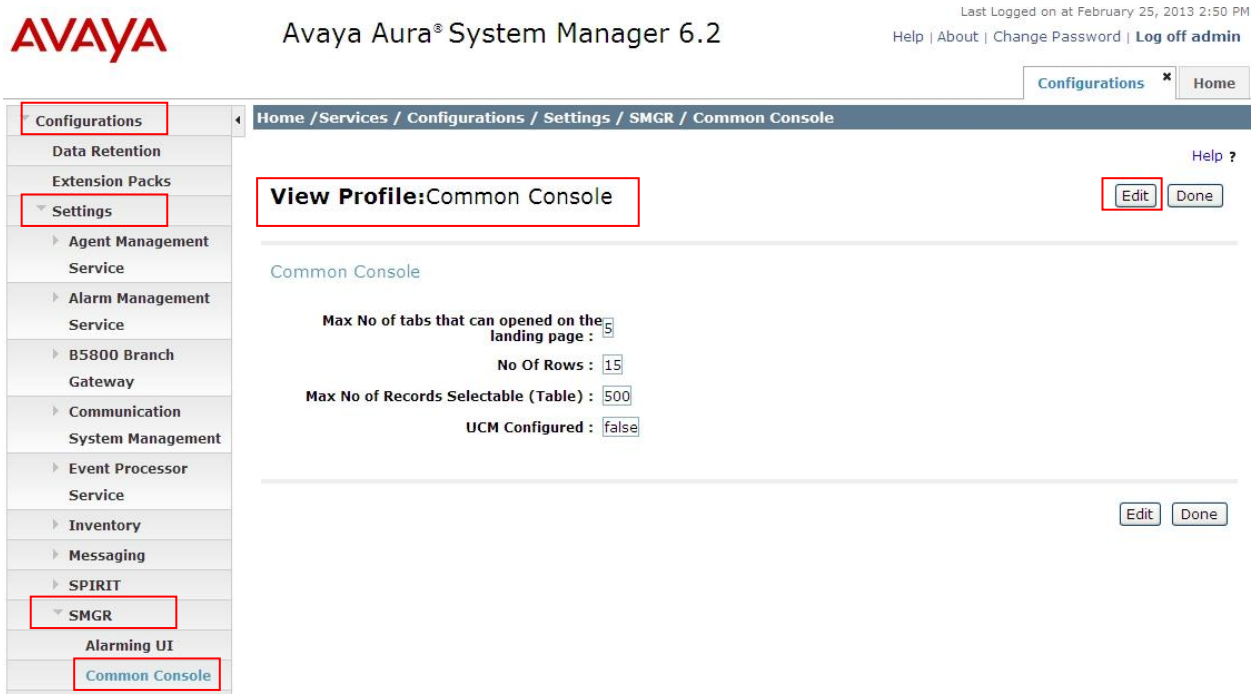
5.2. Enable Avaya Unified Communication Manager services in Avaya Aura® System Manager

A link to UCM Services is not presented by default in System Manager 6.2 console. To enable this link to appear, under the **Services** list, click on **Configurations**.

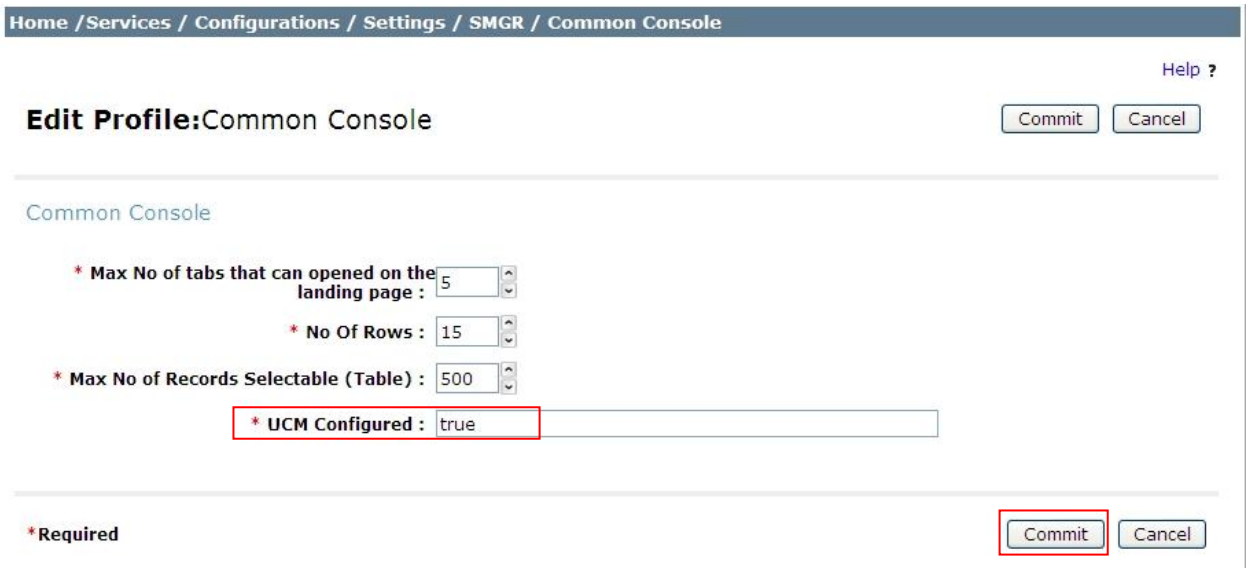


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Under the **Configurations** tab, expand the **Settings** list and then expand the **SMGR** list. Click on the link **Common Console**. In the **View Profile: Common Console** window that appears, select the **Edit** button.

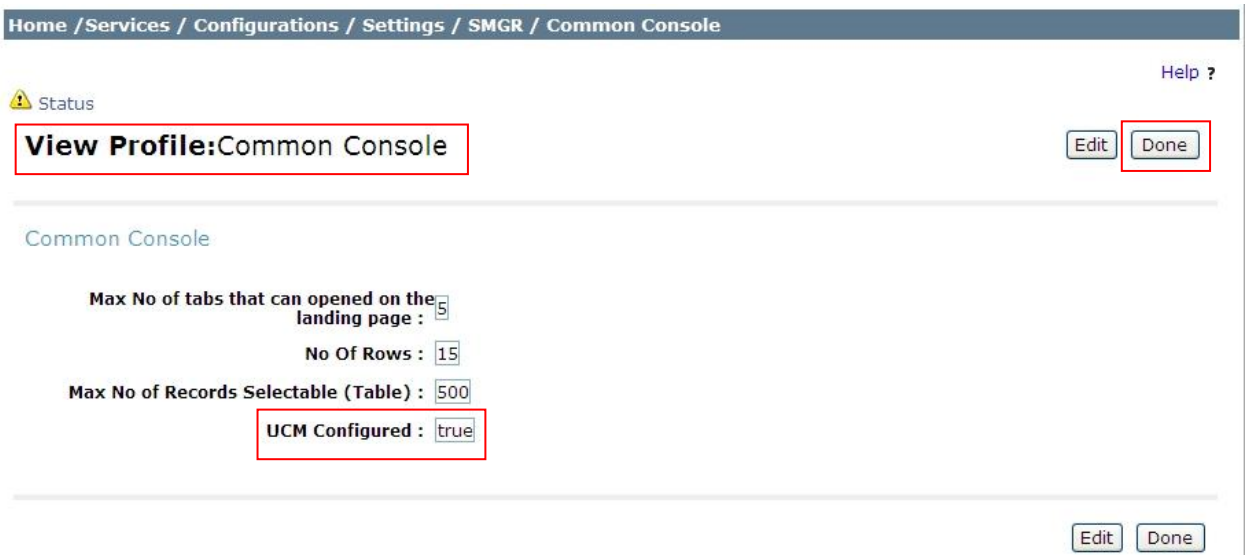


In the **Edit Profile: Common Console** window, change the text in the **UCM Configured** box from **false** to **true** and select the **Commit** button.

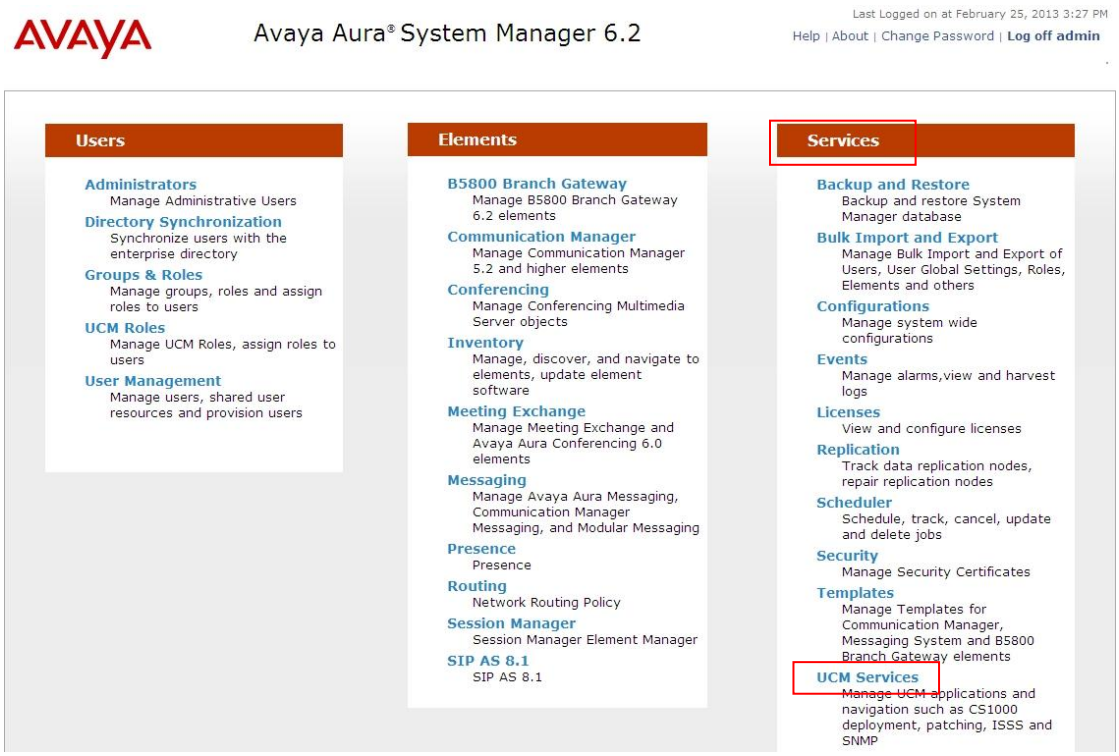


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The change is confirmed in the **View Profile: Common Console** window where **UCM Configured** now shows **true**. Click on **Done** to complete the change.

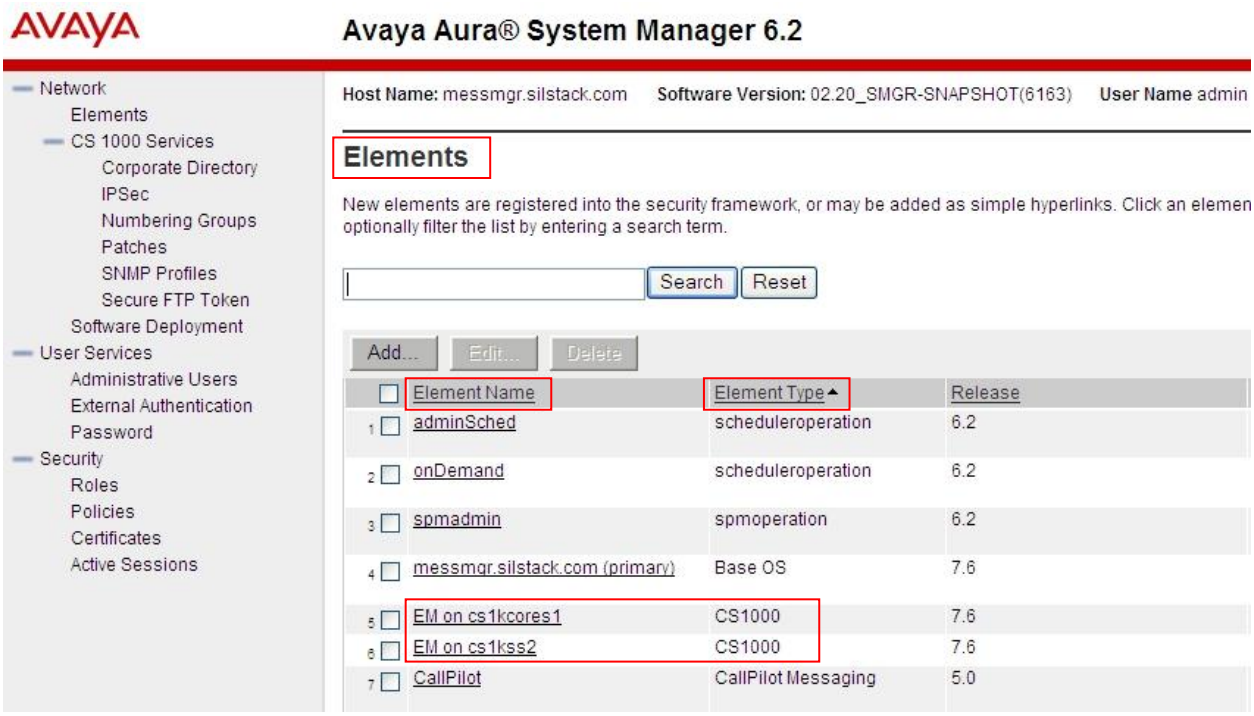


For the change to take effect, logout of System Manager and log back-in. The System Manager Home Page will again be displayed. Under the **Services** category on the right side now select **UCM Services** for CS1000 management.



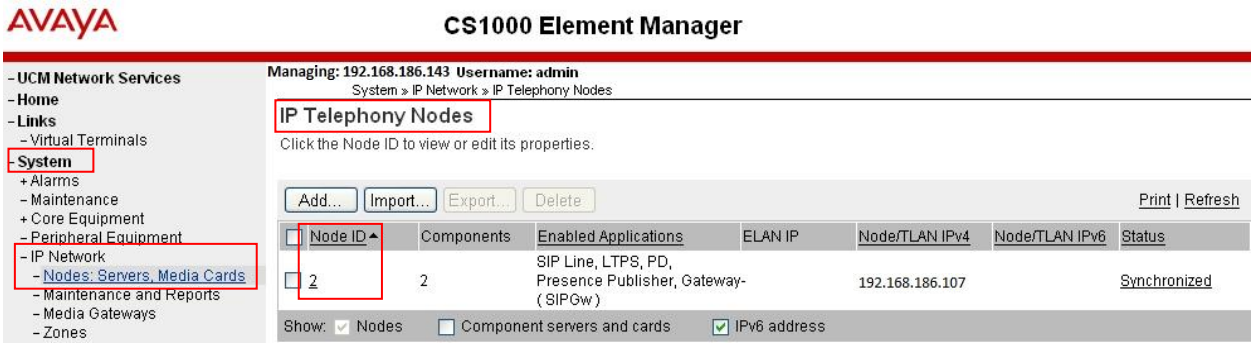
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The UCM **Elements** page opens in a new browser window. Under the **Element Name** column select one of the elements corresponding to **CS1000** in the **Element Type** column (e.g. **EM on cs1kcores1**).



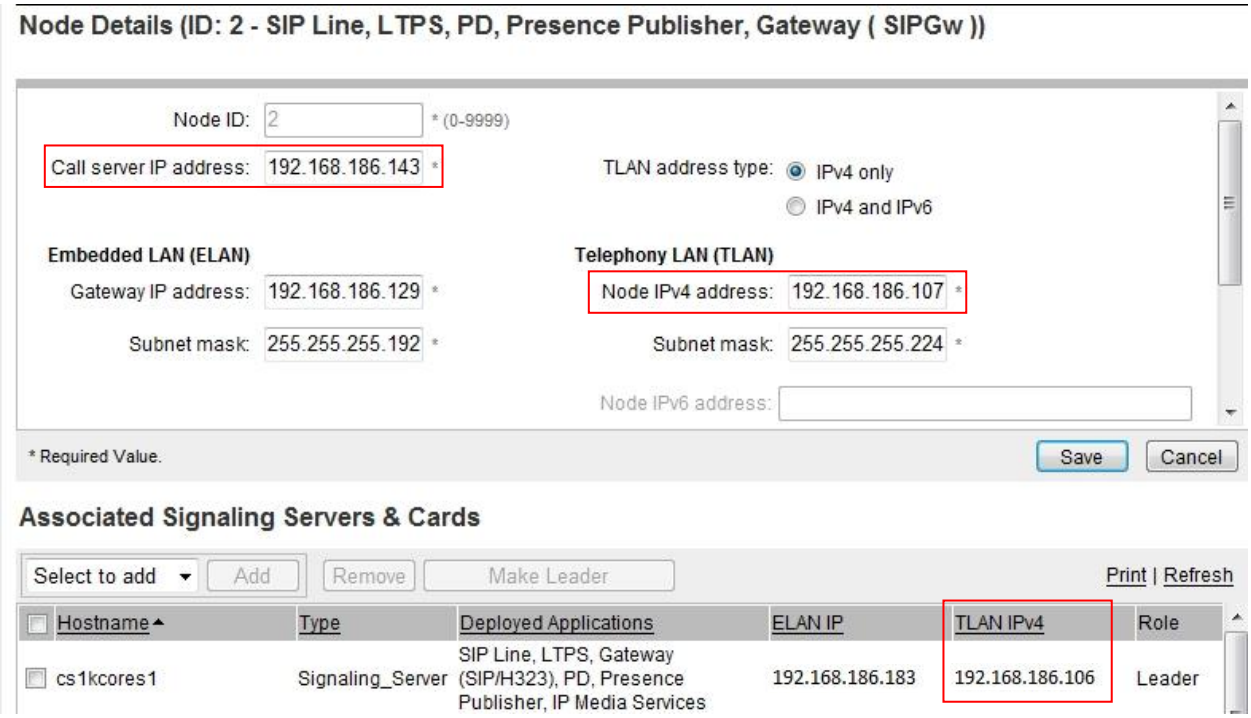
5.3. Confirm Node and IP Addresses

The **CS1000 Element Manager** page appears. On the left hand side, 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 **2**).



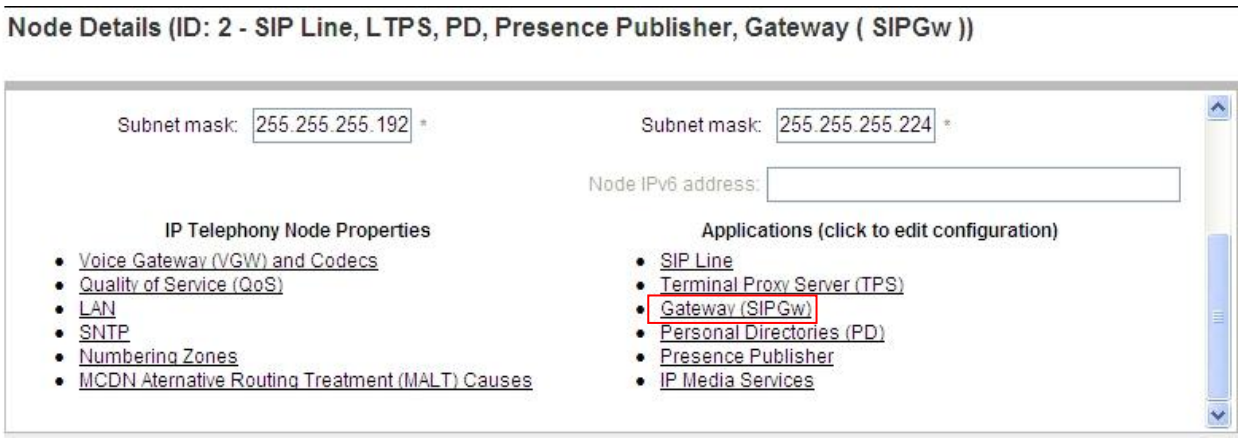
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The **Node Details** screen is displayed with additional details as shown below. Make a note of the **Node IPv4 address, Call server IP address** and **TLAN IPv4** addresses of any Signaling Servers in the node. These addresses are used to configure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



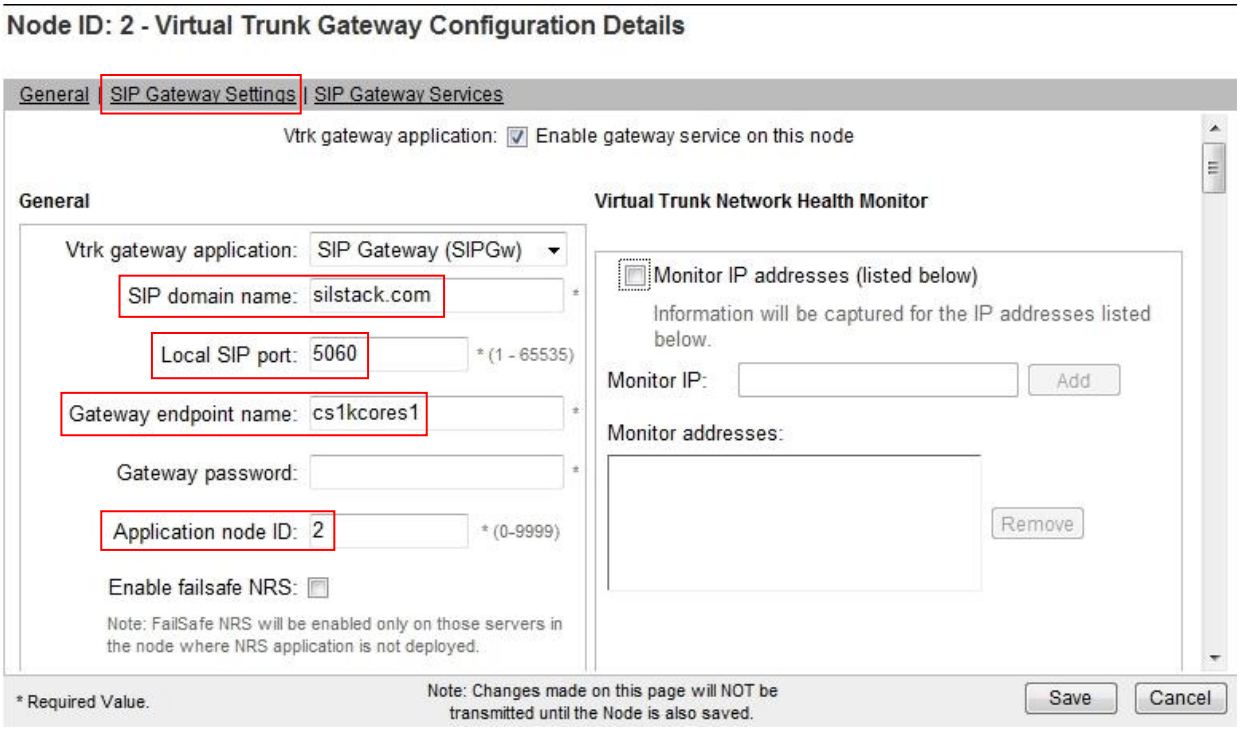
5.4. Configure SIP Trunk to Avaya Aura® Session Manager

While still in the **Node Details** screen as shown above in the previous **section 5.3**, use the scroll bar on the right side of the screen to navigate down to the **Applications** section and select the **Gateway (SIPGw)** link.



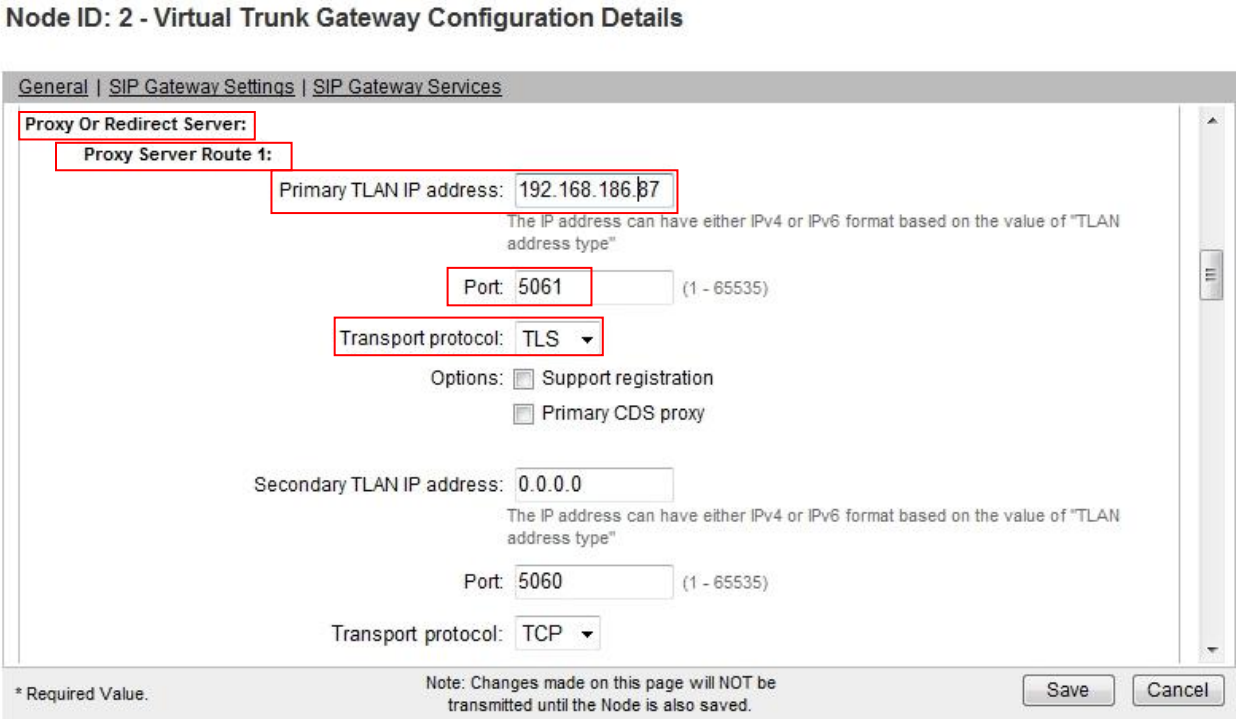
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The **Node ID: 2 - Virtual Trunk Gateway Configuration Details** page appears. Verify that the following fields have been pre-configured or enter new values if not (use default values for fields not specified here). The **SIP domain name** field should contain the SIP domain name for the solution (e.g. **silstack.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. **cs1kcores1**). For the **Application node ID** field enter the Node ID value (e.g. **2**). Then click on the **SIP Gateway Settings** link at the top of this page to jump to that section.



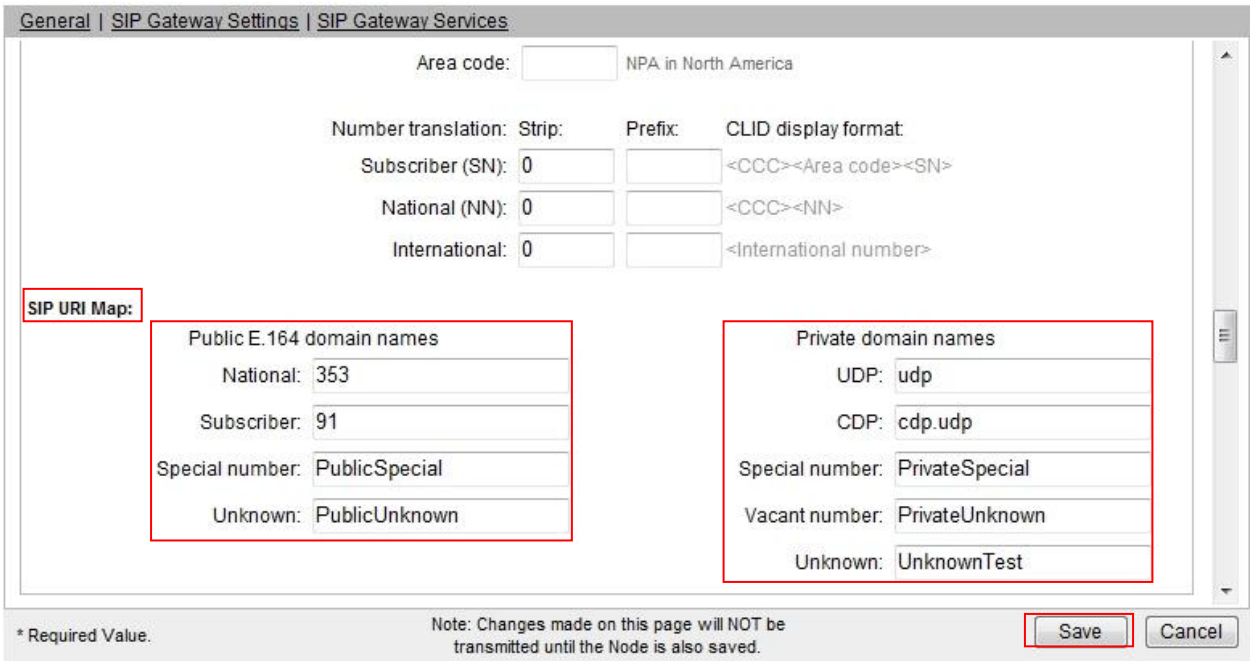
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The SIP Gateway Settings page appears (heading not shown). Scroll down to the **Proxy or Redirect Server** section of the page. In the sub-section titled **Proxy Server Route 1**, enter the IP address of the Session Manager SIP signaling asset in the **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 11**.



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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.

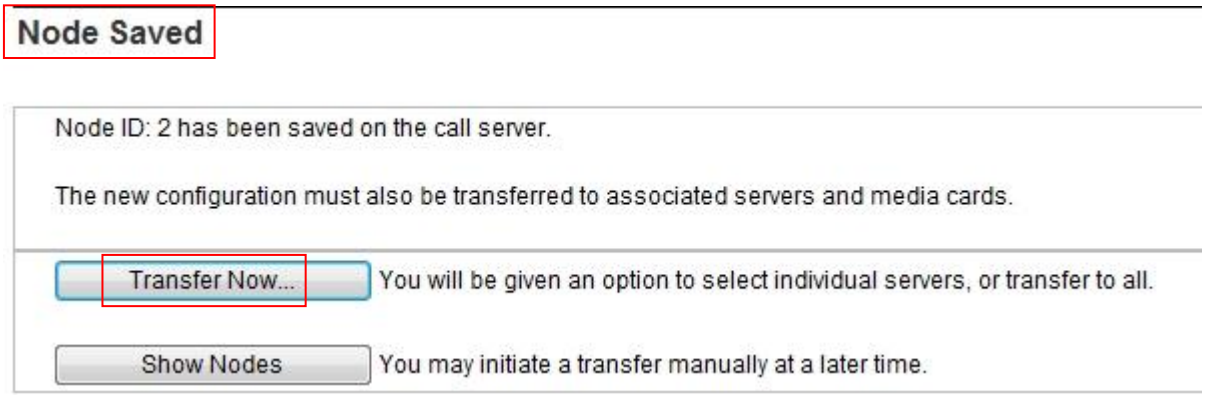


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The **Node ID: 2 - Virtual Trunk Gateway Configuration Details** page re-appears. Press **Save**.

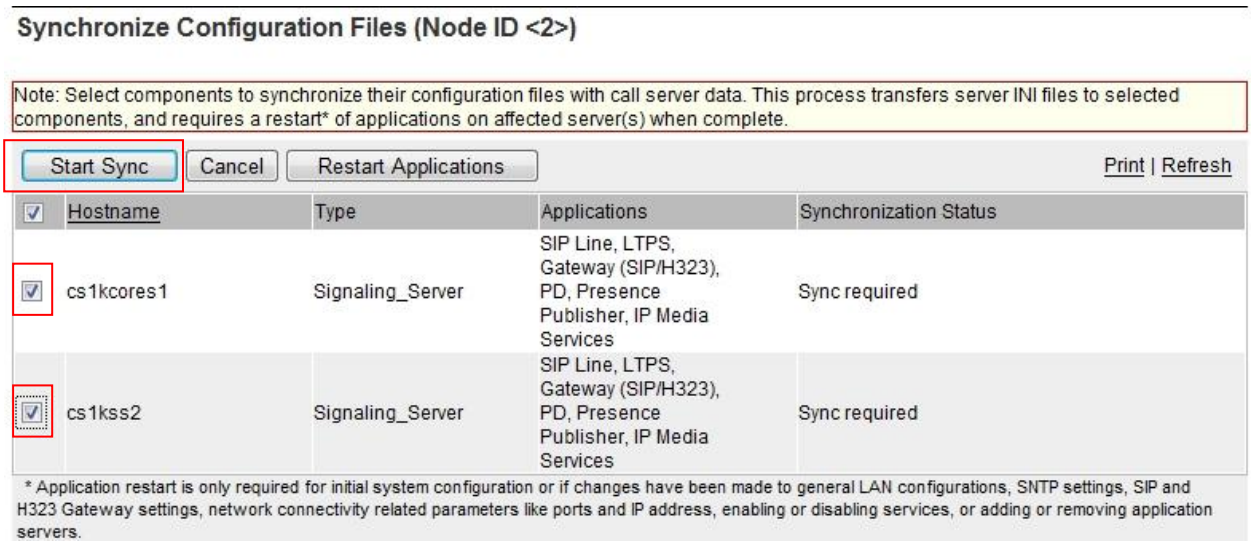


A confirmation message of **Node Saved** appears to indicate that the changes have been saved on the CS1000 Call Server. Select **Transfer Now**.



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Once the transfer is complete, the **Synchronize Configuration Files (Node ID <2>)** page appears. Select all the Signaling Servers listed on this page and click on **Start Sync**.



Synchronize Configuration Files (Node ID <2>)

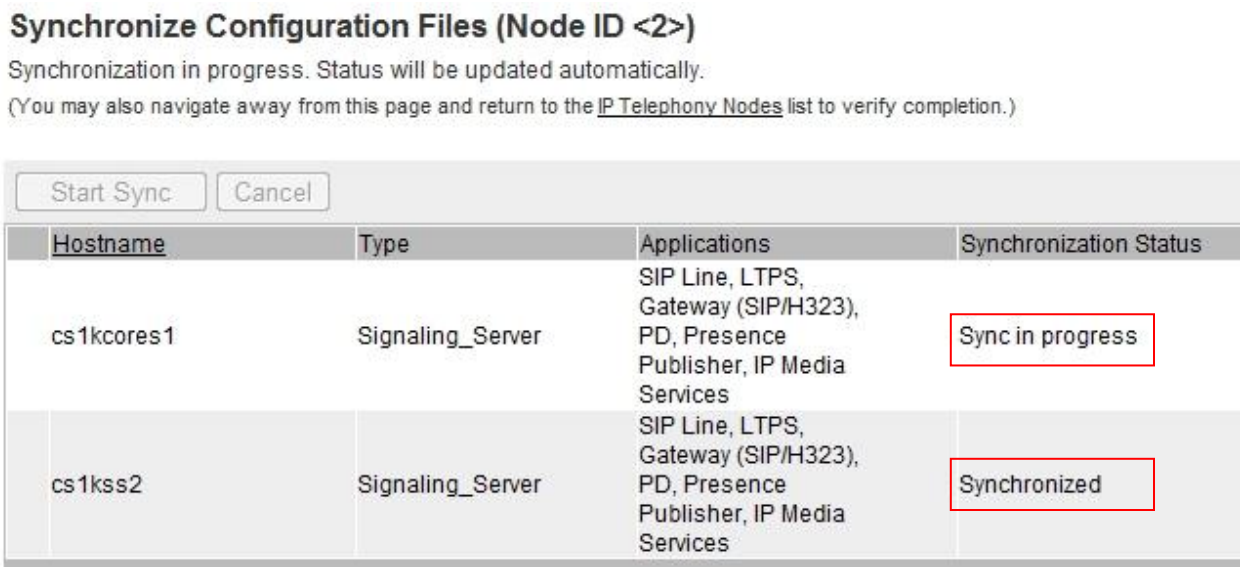
Note: Select components to synchronize their configuration files with call server data. This process transfers server INI files to selected components, and requires a restart* of applications on affected server(s) when complete.

Start Sync | Cancel | Restart Applications | Print | Refresh

	Hostname	Type	Applications	Synchronization Status
☑	cs1kcores1	Signaling_Server	SIP Line, LTPS, Gateway (SIP/H323), PD, Presence Publisher, IP Media Services	Sync required
☑	cs1kss2	Signaling_Server	SIP Line, LTPS, Gateway (SIP/H323), PD, Presence Publisher, IP Media Services	Sync required

* Application restart is only required for initial system configuration or if changes have been made to general LAN configurations, SNTP settings, SIP and H323 Gateway settings, network connectivity related parameters like ports and IP address, enabling or disabling services, or adding or removing application servers.

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).



Synchronize Configuration Files (Node ID <2>)

Synchronization in progress. Status will be updated automatically.

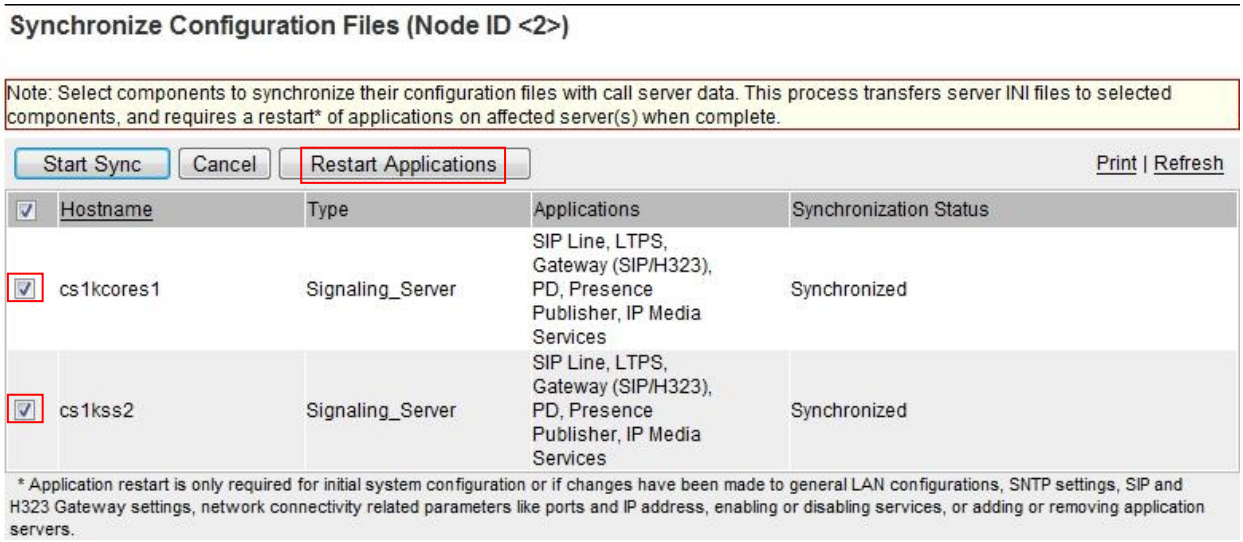
(You may also navigate away from this page and return to the IP Telephony Nodes list to verify completion.)

Start Sync | Cancel

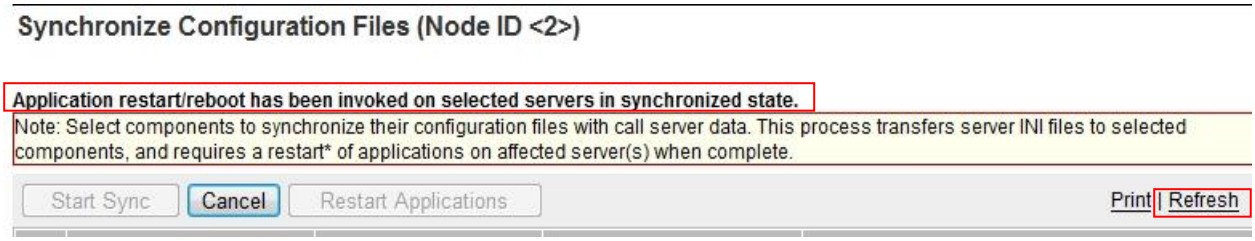
Hostname	Type	Applications	Synchronization Status
cs1kcores1	Signaling_Server	SIP Line, LTPS, Gateway (SIP/H323), PD, Presence Publisher, IP Media Services	Sync in progress
cs1kss2	Signaling_Server	SIP Line, LTPS, Gateway (SIP/H323), PD, Presence Publisher, IP Media Services	Synchronized

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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.



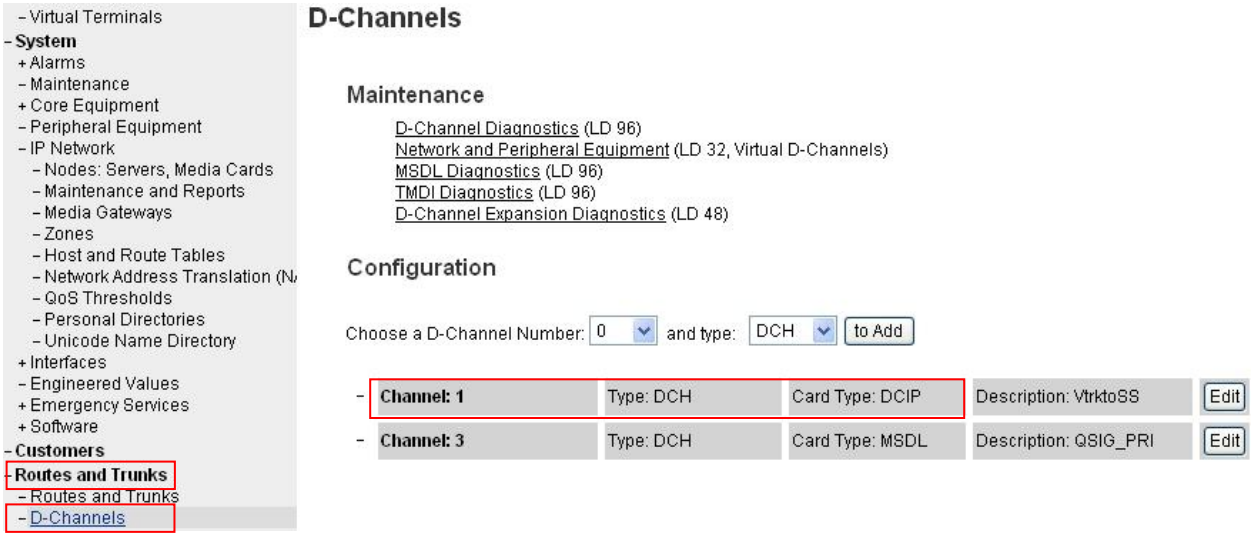
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5.5. Confirm Virtual D-Channel, Routes and Trunks

The CS1000E 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

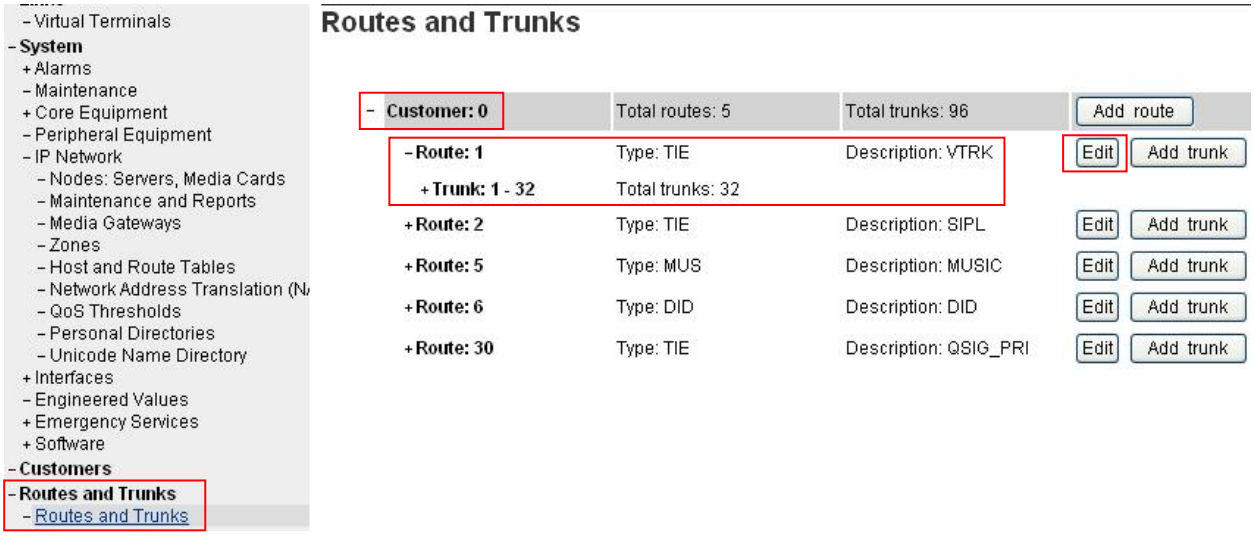
Still in the CS1000 Element Manager page, expand **Routes and Trunks** list on the left navigation panel and select **D-Channels**. The resulting screen shows all the D-channels configured on the CS1000 system. In the sample configuration, there is a single D-channel assigned to **Channel: 1** with **Card Type: DCIP**. Specifying **DCIP** as the card type indicates that the D-channel is a virtual D-channel (i.e. D-Channel over IP).



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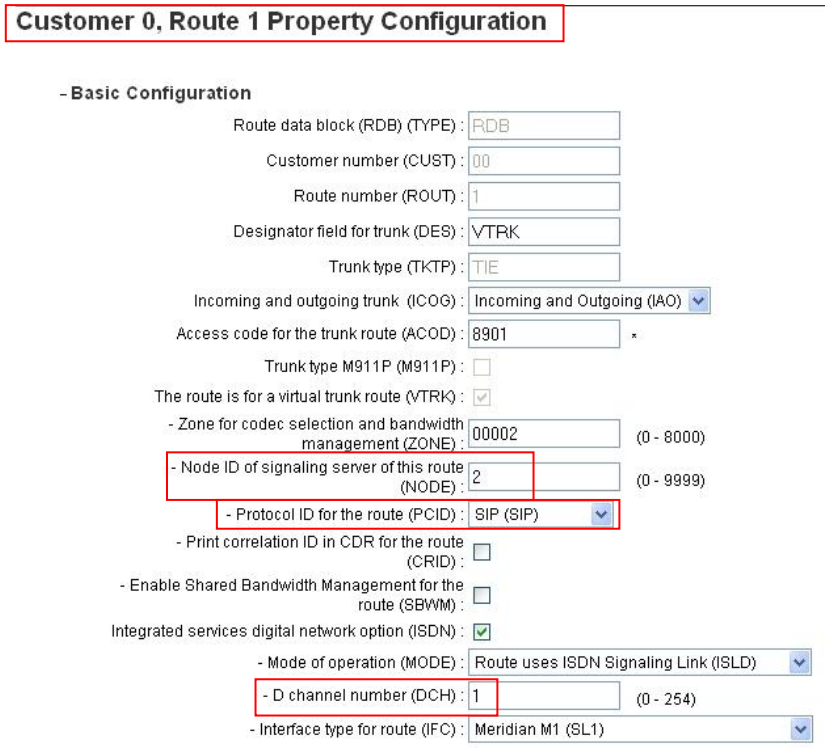
5.5.2. Confirm SIP Route and Trunk Configuration

Still in the CS1000 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 1** is shown with a description of **VTRK** (for Virtual Trunk). Expand **Route 1** 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 1.



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The details of the virtual Route 1 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 **2** and the **Protocol ID for the route (PCID)** has already been set to S**IP (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 5.5.1**.



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5.6. Configure ESN for Route List Index and Digit Manipulation

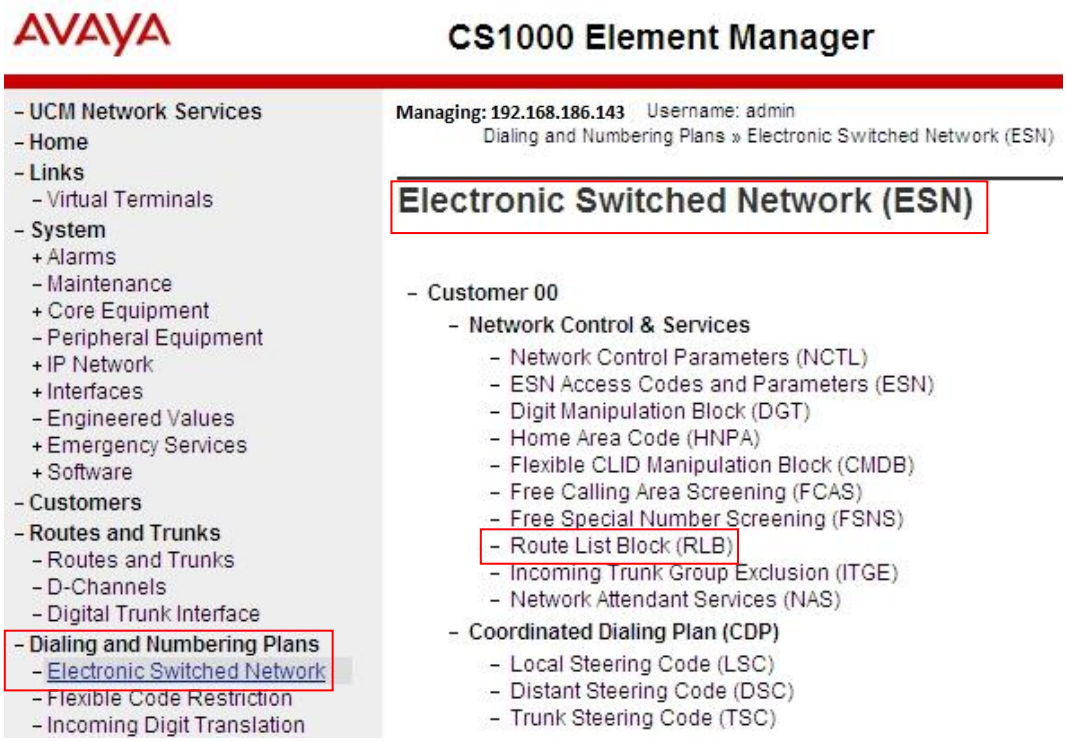
This section provides details of the routing configuration used in the sample configuration to route calls over the SIP Trunk from CS1000 to Session Manager on the Collaboration Pack.

Note: The CS1000 Dialing plan and Collaboration Pack 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 CS1000 endpoints in the sample configuration used a 5-digit number plan with all DNs beginning with 7 (e.g. 7xxxx). All Collaboration Pack endpoints in the sample configuration used a 7-digit number plan with all DNs beginning with 447 (e.g. 447xxxx).

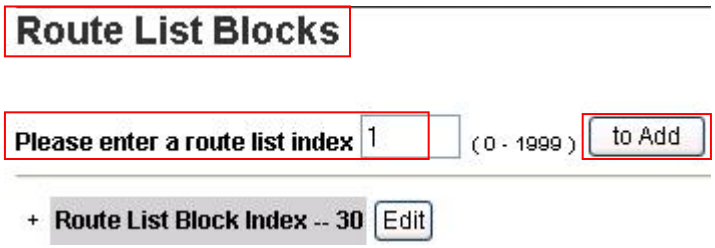
5.6.1. Create Route List Index

From the **CS1000 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)**.

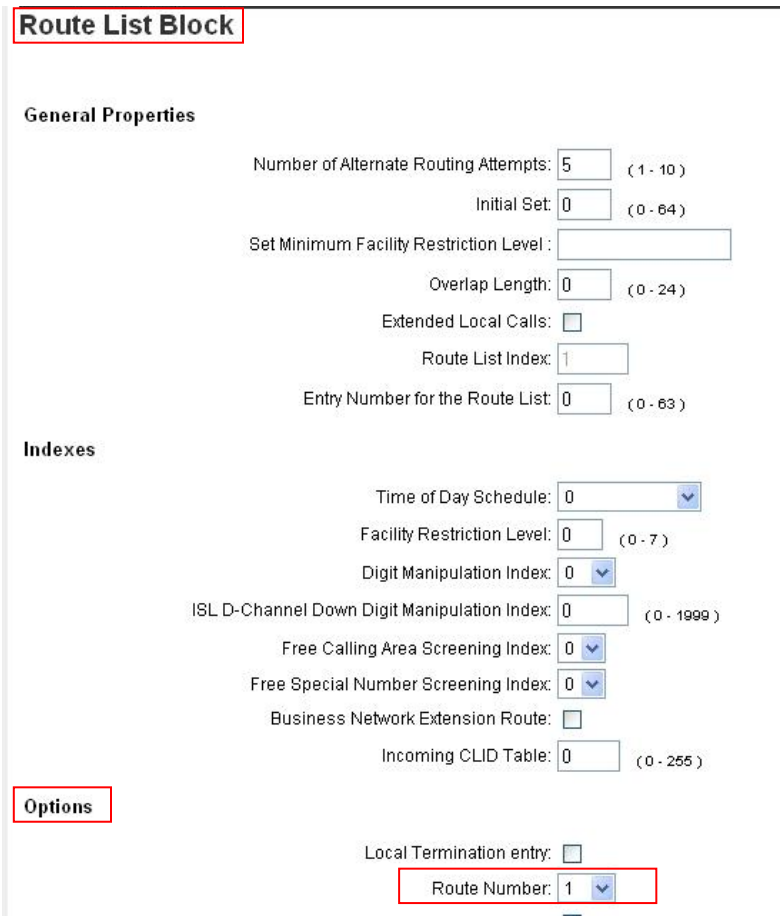


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The **Route List Blocks** screen is displayed. Enter an available route list index number in the **Please enter a route list index** field (e.g. **1**) and click **to Add** as shown below.

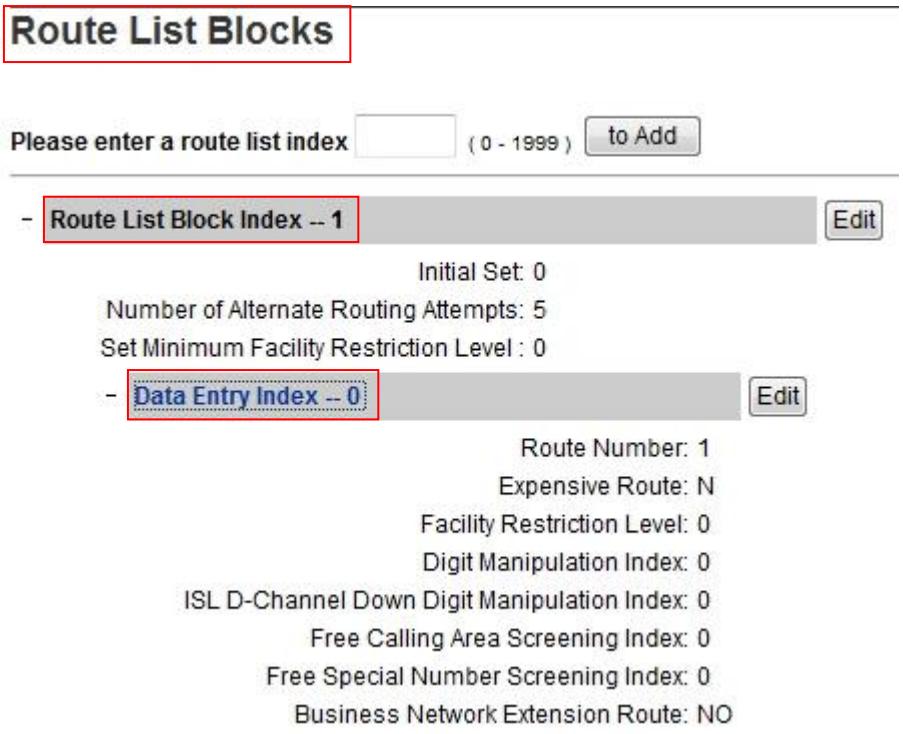


The **Route List Block** window appears. Under the **Options** section, in the drop-down list, select the **Route Number** of the route identified above in **Section 5.6.1** (i.e. Route Number **1**) and use default values for remaining fields as shown below.



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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 -- 1** 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 CS1000 user's Personal Call Assistant (PCA) configuration to route voice calls to the Collaboration Pack via the Session Manager. A different DSC will be used specifically to route Message Waiting Indicator (MWI) messages from CallPilot across the SIP trunk to Collaboration Pack via the Session Manager. This is because non-call associated messages such as MWI are treated different to regular voice calls by Session Manager and Communication Manager so it needs to be able to distinguish them. The following table summarizes this as follows:

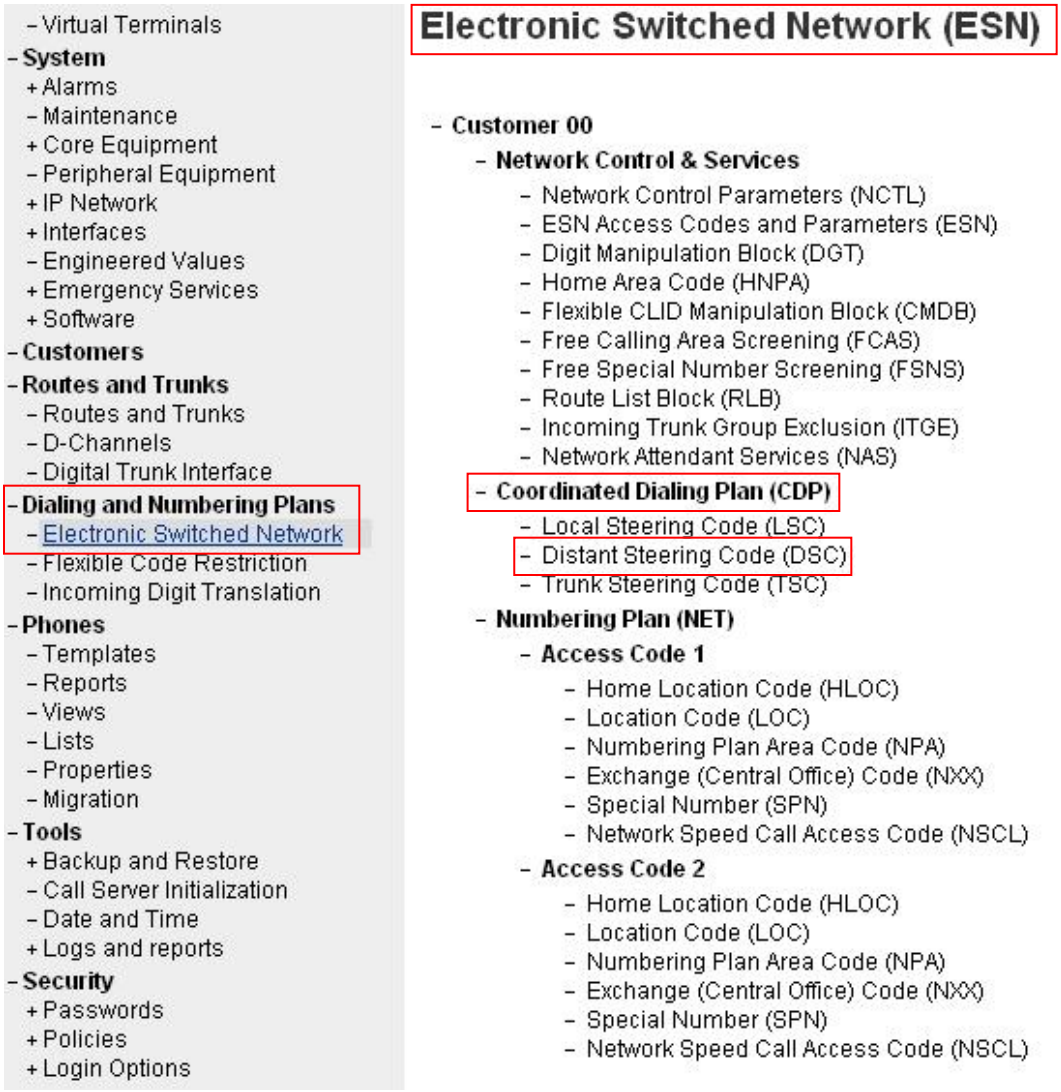
Number type	Prefix (DSC) / FLEN	RLI	Route
PCA: 88447xxxx	884 / 9	1	1
MWI: 447xxxx	447 / 7	1	1

So for example, a call made to CS1000 set DN 70015 will also get sent, using the PCA feature, to DN 884470015 (which is a 9 digit DN). DSC 884, which has a Flexible Length (FLEN) of 9, will send these 9 digits via RLI 1 to route 1 which is the SIP trunk route to Collaboration Pack. 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 CS1000 and also to 4470015 (this number is programmed in

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CallPilot configuration for this user's mailbox). DSC 447 (using FLEN of 7 digits) will send this number via RLI 1 to route 1 which again is the SIP trunk route to Collaboration Pack. So the difference is that for the voice call, the number sent to Collaboration Pack will be 884470015 while for MWI, the number sent will be 4470015.

From the **CS1000 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.

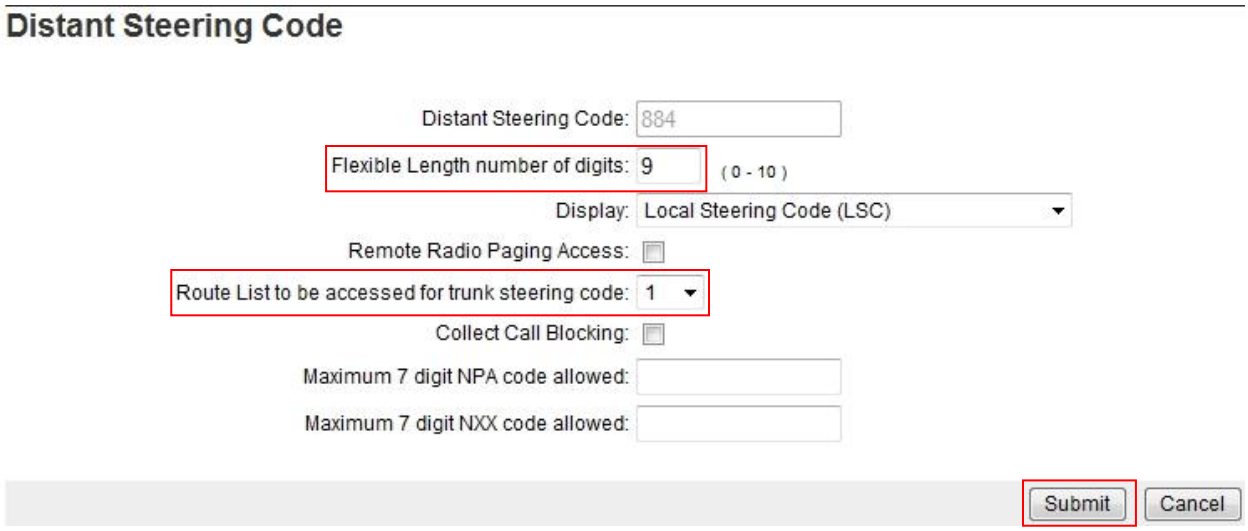


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On the **Distant Steering Code List** page, select **Add** from the drop-down menu. In the box titled **Please enter a distant steering code** enter the dialed prefix for calls to be routed over SIP trunk to Session Manager (e.g. enter **884**). Click **to Add** button to create the new distant steering code.



The **Distant Steering Code** window appears. For **Flexible Length number of digits**, enter **9**. For **Route List to be accessed for trunk steering code**, select the number **1** 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.



A confirmation pop-up message appears to indicate that the DSC was successfully added. Press OK to clear the message box.

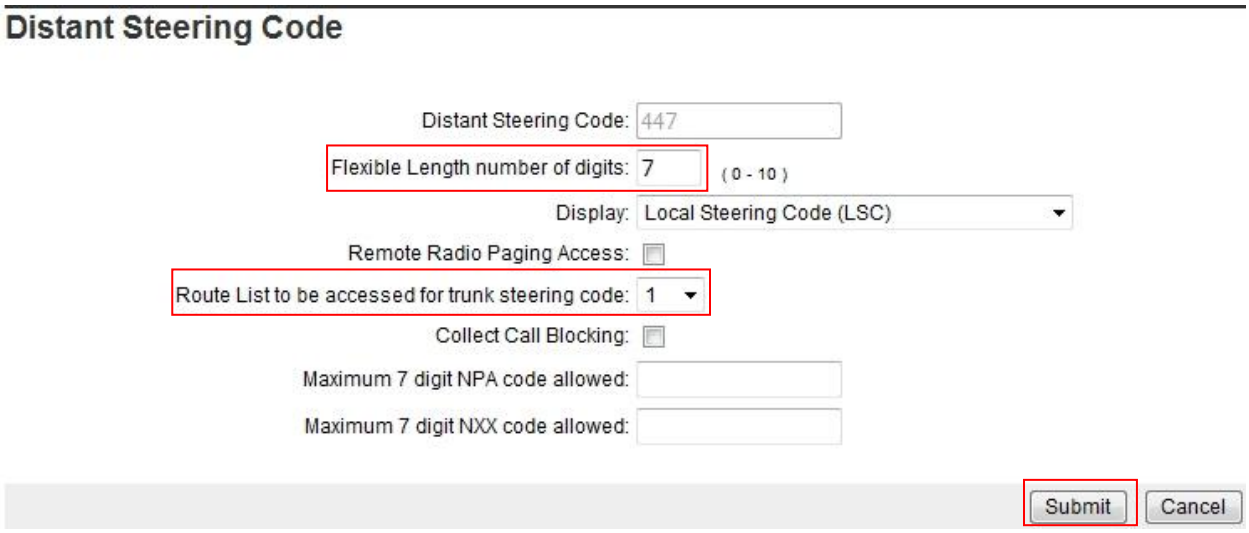


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Repeat the procedure to add a DSC for 447. On the **Distant Steering Code List** page, select **Add** from the drop-down menu. In the box titled **Please enter a distant steering code** enter the dialed prefix for MWI messages to be routed over SIP 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 **1** 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.



As before, a confirmation pop-up message appears to indicate that the DSC was successfully added. Press OK to clear the message box.

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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. **884**) and click on the **View** button. Expand the **Distant Steering Code List – 884** item to show the parameters of the DSC. Repeat the same for DSC 447 to verify it has been successfully programmed (not shown).



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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 CS1000 and the Collaboration Pack.

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 CS1000.

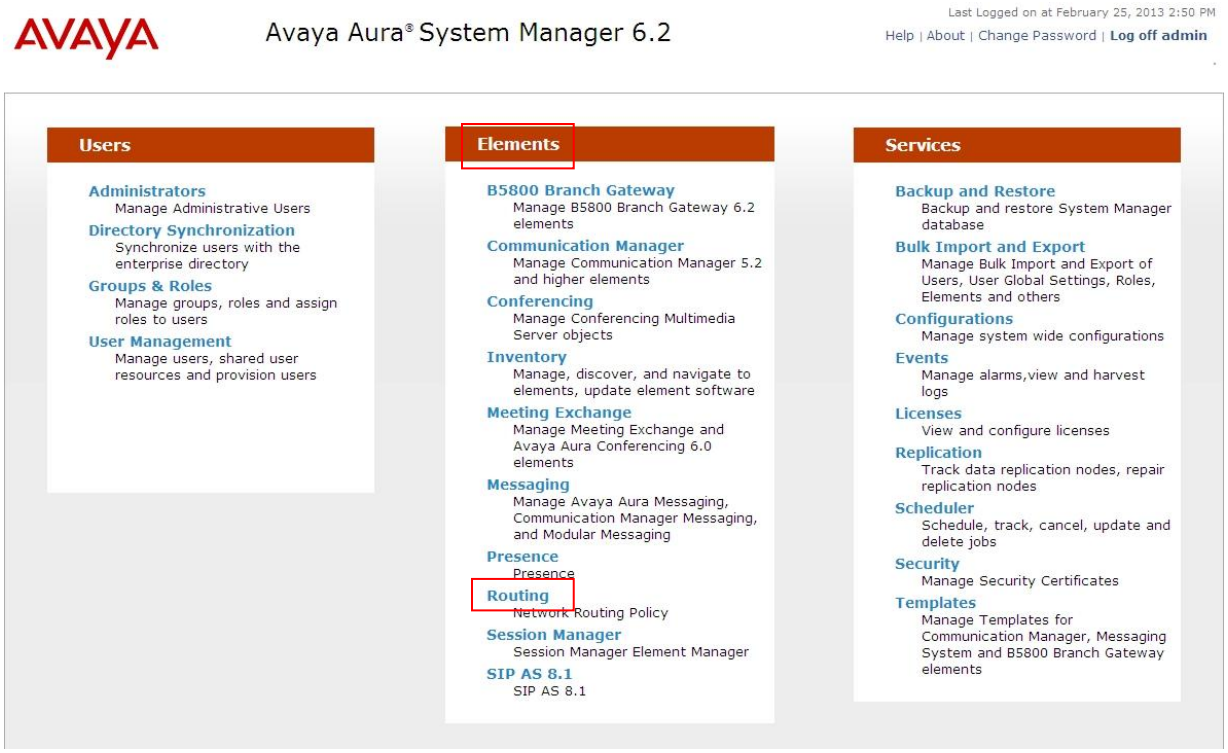
Specifically, the following administration activities will be described:

- Verify SIP Domains
- Define a Location for Avaya Communication Server 1000
- Configure Adaptation Module
- Define SIP Entities
    - SIP Entity for CS1000
    - 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.

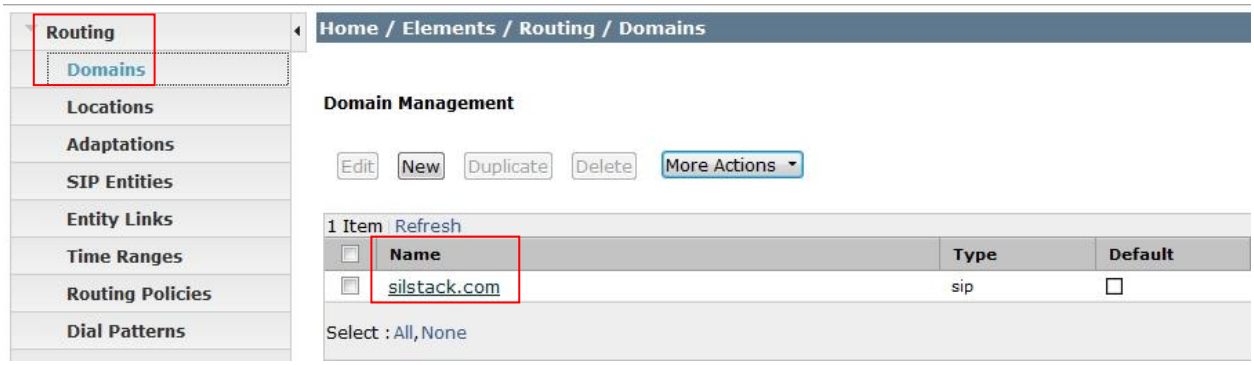
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6.1. Verify SIP Domains

From the main System Manager 6.2 page under **Elements**, click on **Routing**.



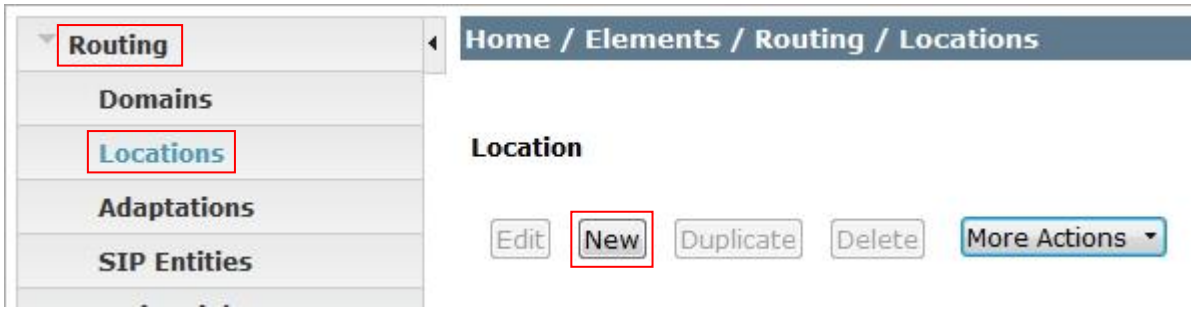
Expand the **Routing** list and select **Domains** from the left navigation menu and verify that the Domain **Name** is the same as the CS1000 SIP Gateway Domain as shown in **Section 5.4**. In this sample configuration **silstack.com** was used.



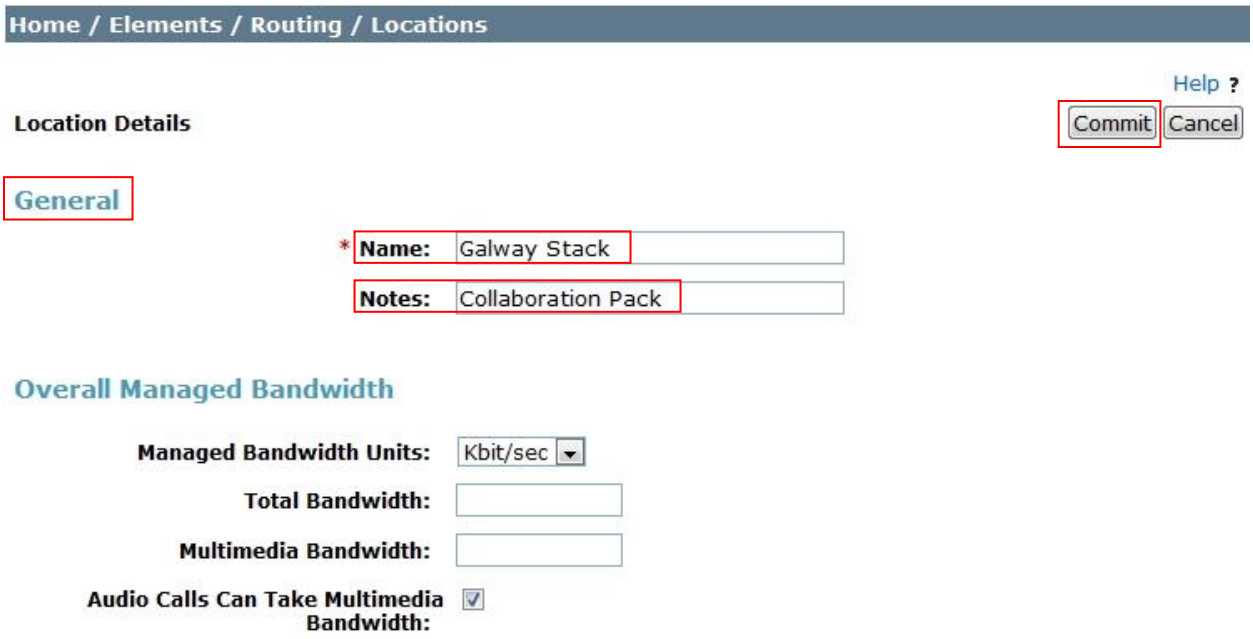
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6.2. Define Location for Avaya Communication Server 1000

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**.

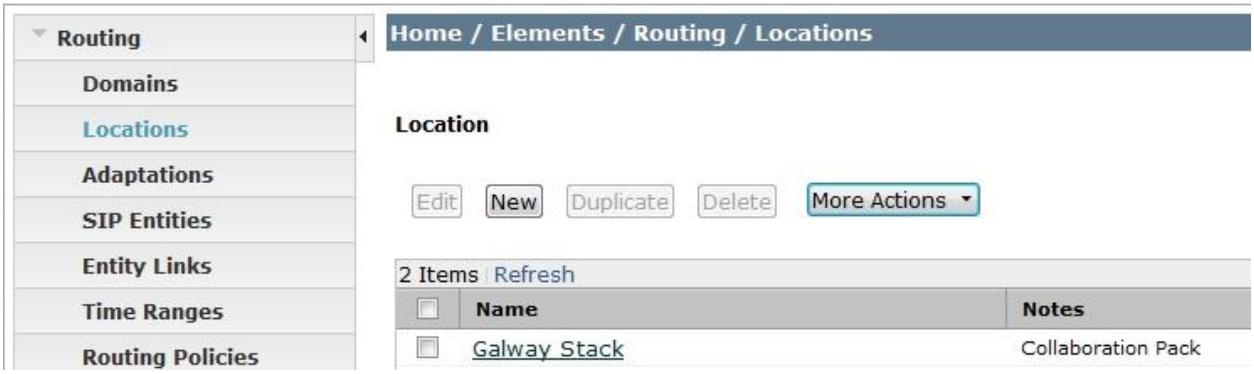


In the **General** section, enter a location name in the **Name** box (e.g. **Galway Stack**). Optionally, enter some text into the **Notes** box (e.g. **Collaboration Pack**). Use the default values for all remaining fields and click **Commit** to save.



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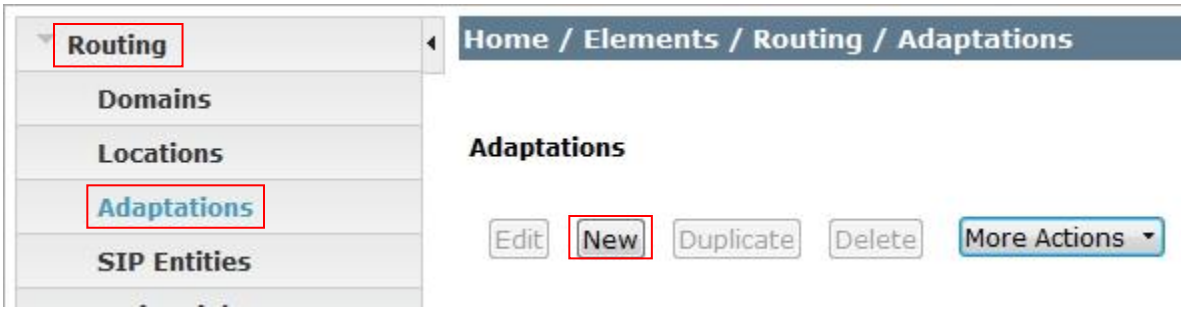
The resultant screen shows the Galway Stack location defined for CS1000 Collaboration Pack in the sample configuration.



6.3. Configure Adaptation Module

To enable calls between endpoints on CS1000 and SIP endpoints registered to Session Manager, Session Manager should be configured to use an Adaptation Module designed for CS1000 to convert SIP headers in messages sent by CS1000 to the format used by other Avaya products and endpoints. All calls to CS1000 will have the CS1000 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 CS1000. It will be necessary to delete the leading '1' for those calls that terminate within CS1000 (station calls and Call Pilot).

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



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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. **CS1000**).
- **Module Name:** Select **CS1000Adapter** 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.

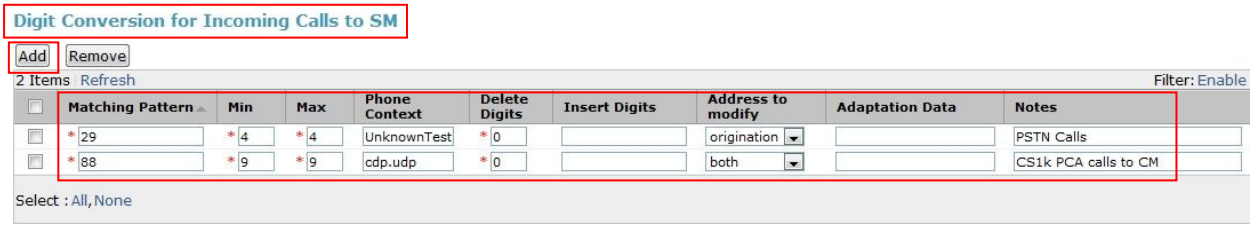


Matching patterns will be used for certain calls to adapt the SIP phone context information for the relevant call. In the sample configuration to simulate a PSTN call, an ISDN trunk was configured between the CS1000E and a second CS1000E system where a set with DN 2997 was used as the PSTN endpoint. 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 CS1000 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 CS1000 to a Collaborated endpoint the call is extended over SIP using a PCA and a CDP DSC of 884. Therefore a matching pattern of 88 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
88	9	9	cdp.udp	0		both	CS1k PCA calls to CM

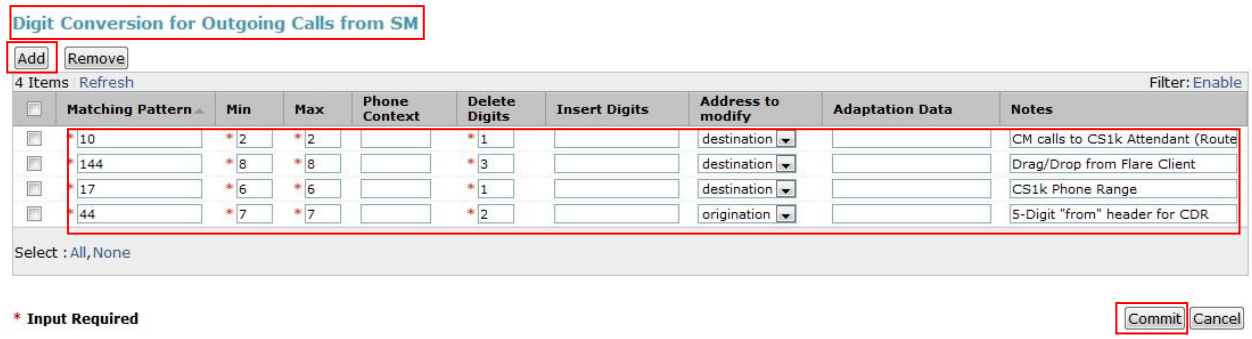
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In the **Digit Conversion for Incoming Calls to SM** section, click **Add** and enter these values. It is also recommended to enter optional text in the **Notes** section as a description for each entry.



In the **Digit Conversion for Outgoing Calls from SM** section, click **Add** and enter appropriate values for the Customer solution. These represent calls from the Collaboration Pack to CS1000 where the SIP addresses (e.g. destination address) have to be modified for proper interoperability. For example in the sample configuration, all calls from Collaborated endpoints to CS1000 sets have a prefix of 1 inserted as a means of directing those calls to CS1000. The 5-digit DNs for all sets on the CS1000 start with digit 7, so a matching pattern of 17 is required to delete the prefix 1 before the destination number is passed to CS1000. The values shown in the table below were used in the sample configuration.

Matching Pattern	Min	Max	Delete Digits	Insert Digits	Address to modify	Notes
10	2	2	1		destination	CM calls to CS1k Attendant
144	8	8	3		destination	Drag/Drop from Flare Client
17	6	6	1		destination	CS1k Phone Range
44	7	7	2		origination	5-Digit "from" header for CDR



Click **Commit** to save the changes.

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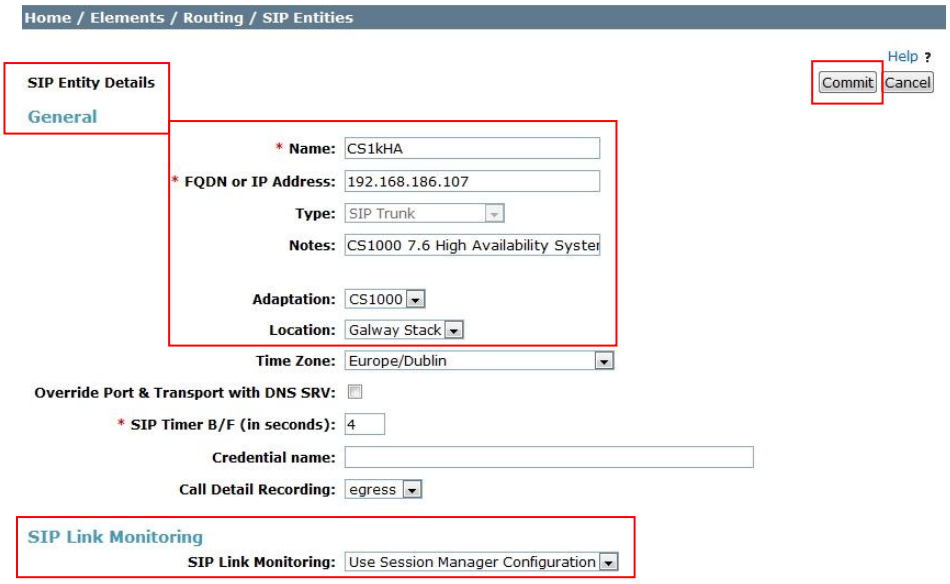
6.4. Define SIP Entities

SIP entities must next be added for CS1000 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 CS1000

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 CS1000 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. **CS1000 7.6 High Availability System**). From the drop-down **Adaptation** list select the **CS1000** Adaptation Module defined earlier in **Section 6.3**. From the drop-down **Location** list, select the **Galway Stack** Location for CS1000 as defined earlier in **Section 6.2**. 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.



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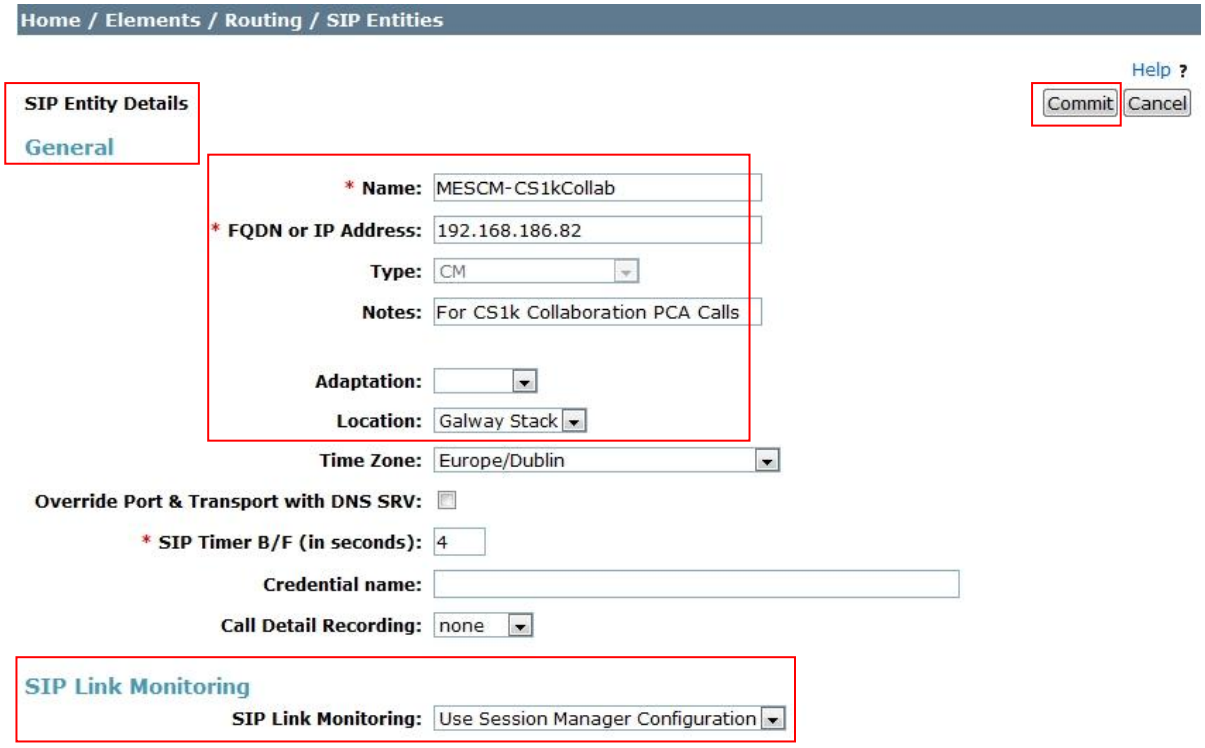
A second SIP Entity is required for the CS1000 to be associated with the TLS link required to send presence updates from CS1000 to the Presence server. The name will be different than the first Entity created above and no adaptation will be applied. **Note**: configuration details required for Presence Services are not covered in this Application Note – check references in **Section 11** 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 CS1000 and calls to PSTN via CS1000). 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 6.4**). 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 Collaboration PCA Calls**). From the drop-down **Location** list, select the **Galway Stack** Location as defined earlier in **Section 6.2**. 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.

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6.5. Define Entity Links

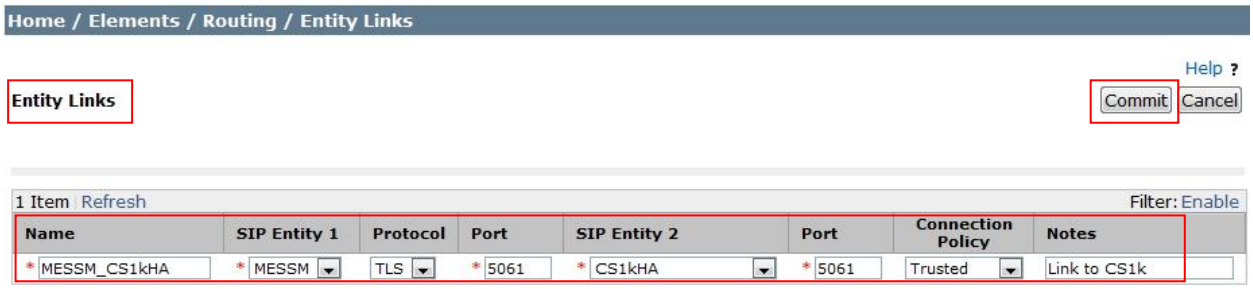
Any connections to Session Manager are described by an Entity Link. In the sample configuration there are two Entity Links between Session Manager and CS1000, one for MWI/Voice traffic and one for Presence Services. Likewise, two Entity links are required between Session Manager and Communication Manager, one for Collaboration client IMS services (created during the ME server installation) and one for communication with the CS1000. 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.



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6.5.1. Entity Link for Avaya Communication Server 1000

The SIP trunk between Session Manager and CS1000 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 CS1000 for voice calls in **Section 6.4** (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.

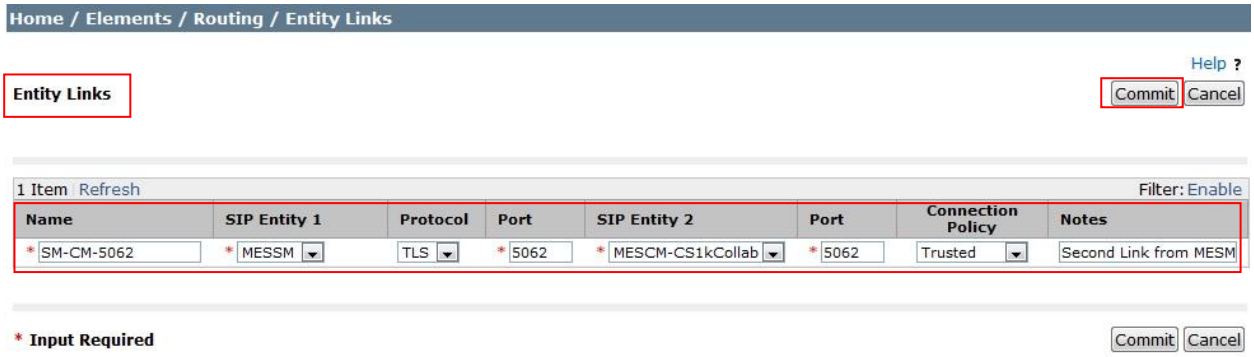


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6.5.2. Entity Link for Communication Manager

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 CS1000 and calls to PSTN via CS1000). 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.

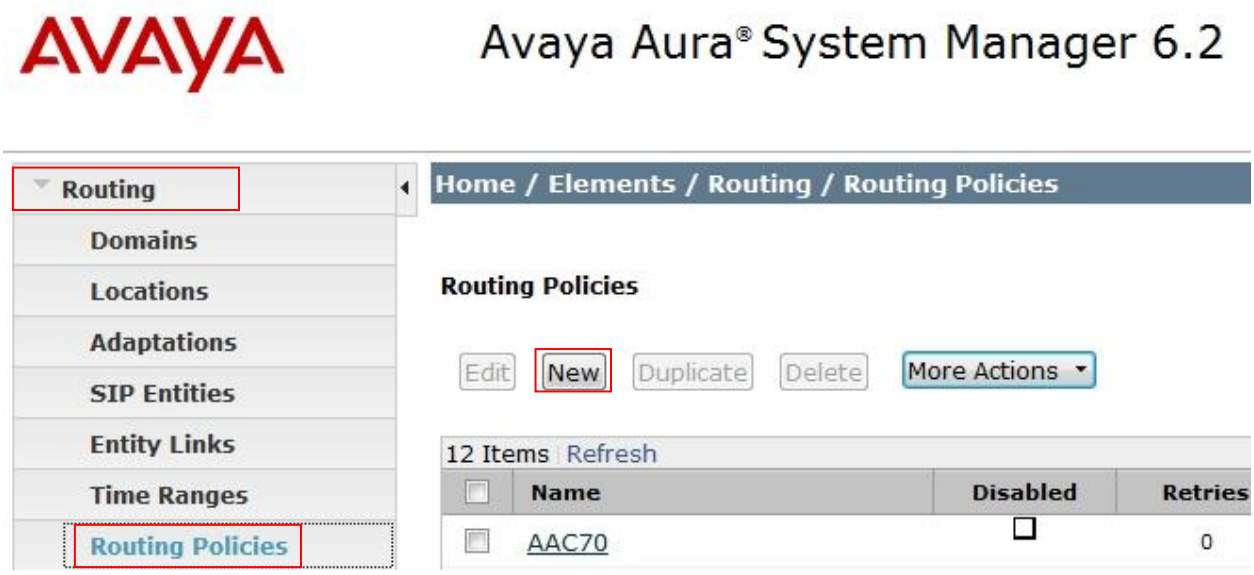
From the System Manager main page under the Routing tab, select Entity Links from the left navigational menu and click New to create a new Entity Link (as shown above in **Section 6.5**). In the **Entity Links** page, for **Name**, enter an identifier for the link to each telephony system (e.g. **SM-CM-5062**). 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 6.4.2** (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.



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6.6. Define Routing Policy

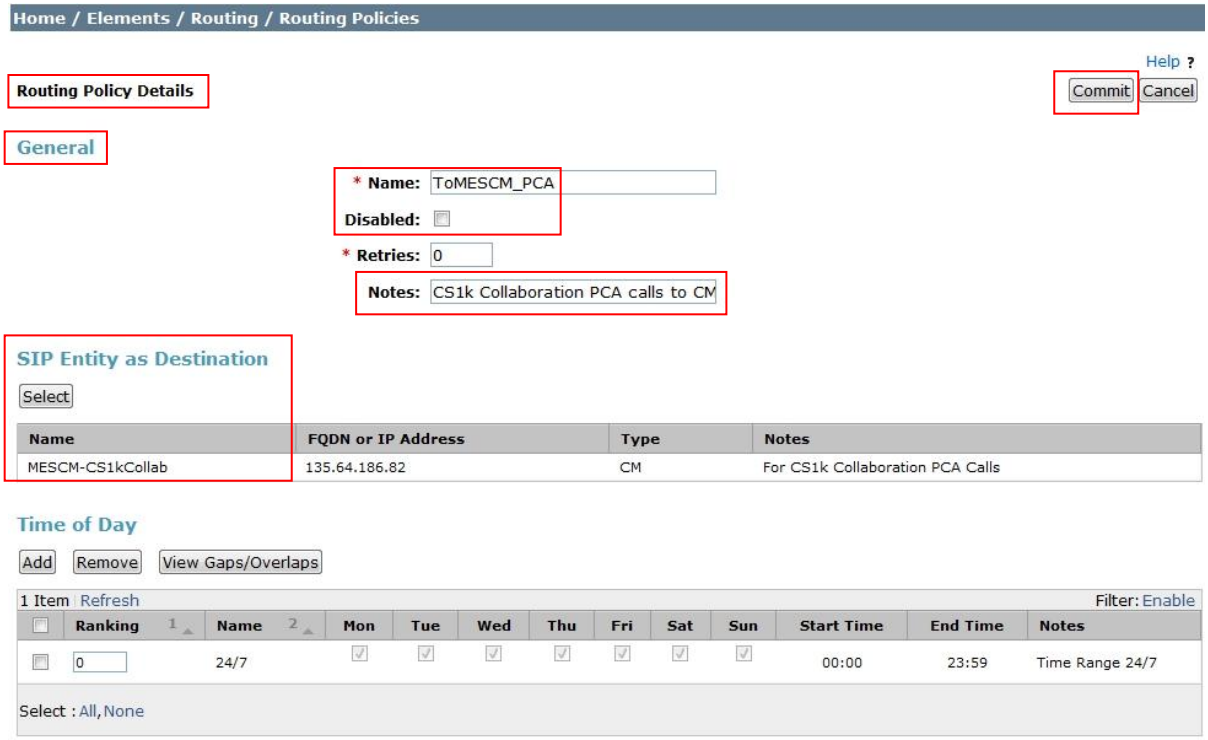
Routing policies describe the conditions under which calls will be routed to CS1000 from SIP endpoints registered to Session Manager. Routing Policies will also be used to send calls and Callpilot MWI notification messages from the CS1000 to Communication Manager. From the System Manager main page under the **Routing** tab, select **Routing Policies** from the left navigational menu. Click **New** to create a new Routing Policy.



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6.6.1. Define the Routing Policy for calls to Avaya Aura® Communication Manager

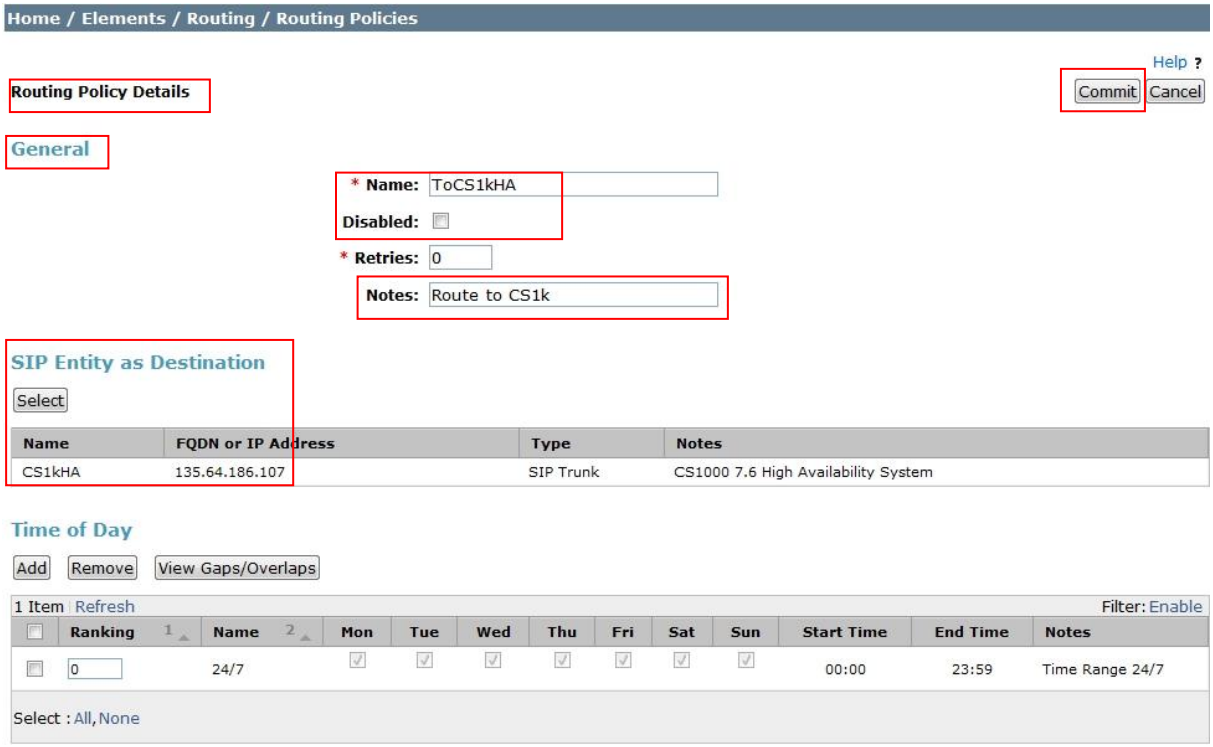
In the **General** section, enter the following values. For **Name**, enter an identifier to define the routing policy (e.g. **ToMESCM_PCA**). Leave the **Disabled** box unchecked. Optionally, enter some descriptive text in the **Notes** box (e.g. **CS1k Collaboration PCA calls to CM**). In the **SIP 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 6.4.2** (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.



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

Following the same procedure as above in **Section 6.6.1**, a Routing Policy is built to CS1000. 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 CS1000 as defined above in **Section 6.4.1** (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.

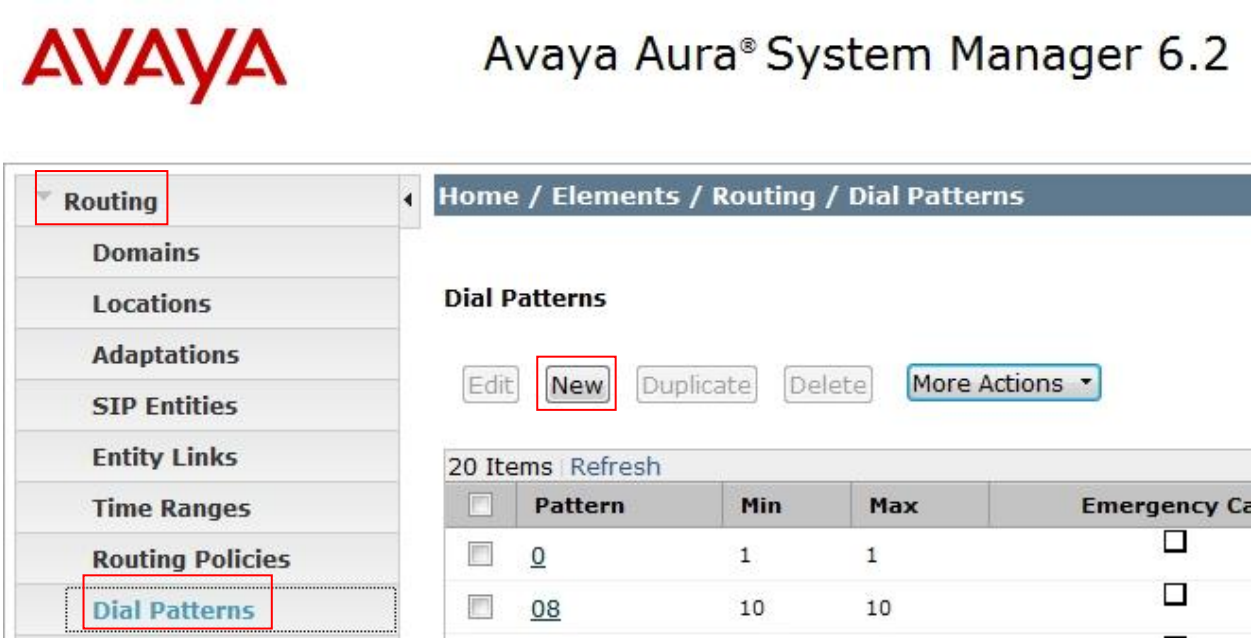


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6.7. Define Dial Pattern

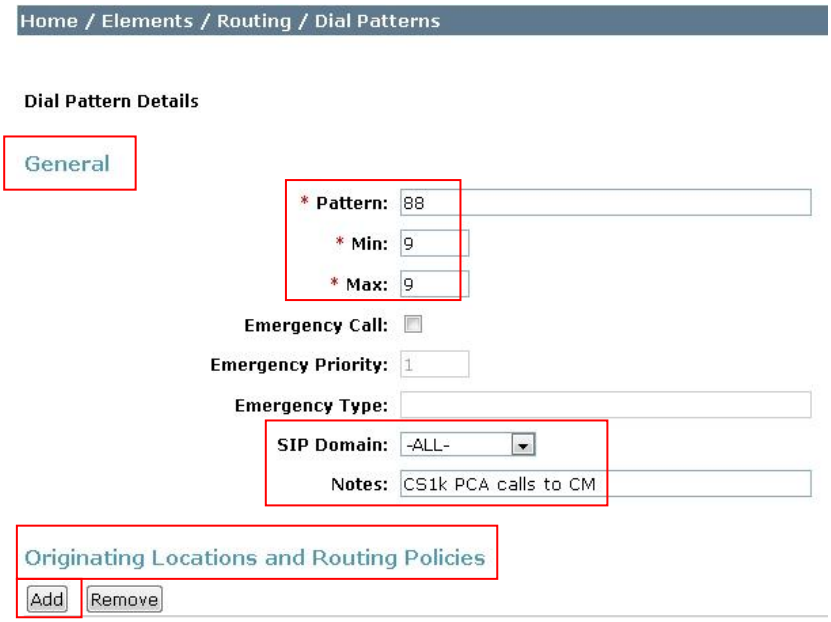
Dial patterns are used to route calls to appropriate SIP Entities. In the sample configuration, voice calls from CS1000 to Communication Manager are extended by PCA with a prefix of 88 in front of the 7 digit DN. So a dial pattern of 88 is used to route these calls to Communication Manager. Calls from the Collaboration pack stations to CS1000 are routed first to Communication Manager per the user profile application sequencing configuration. Communication Manager inserts a prefix digit 1 in front of the 5-digit extension. This is then sent back to Session Manager where a matching dial pattern of 1 routes the call to CS1000. The following text just describes how to configure these two dial patterns. 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.



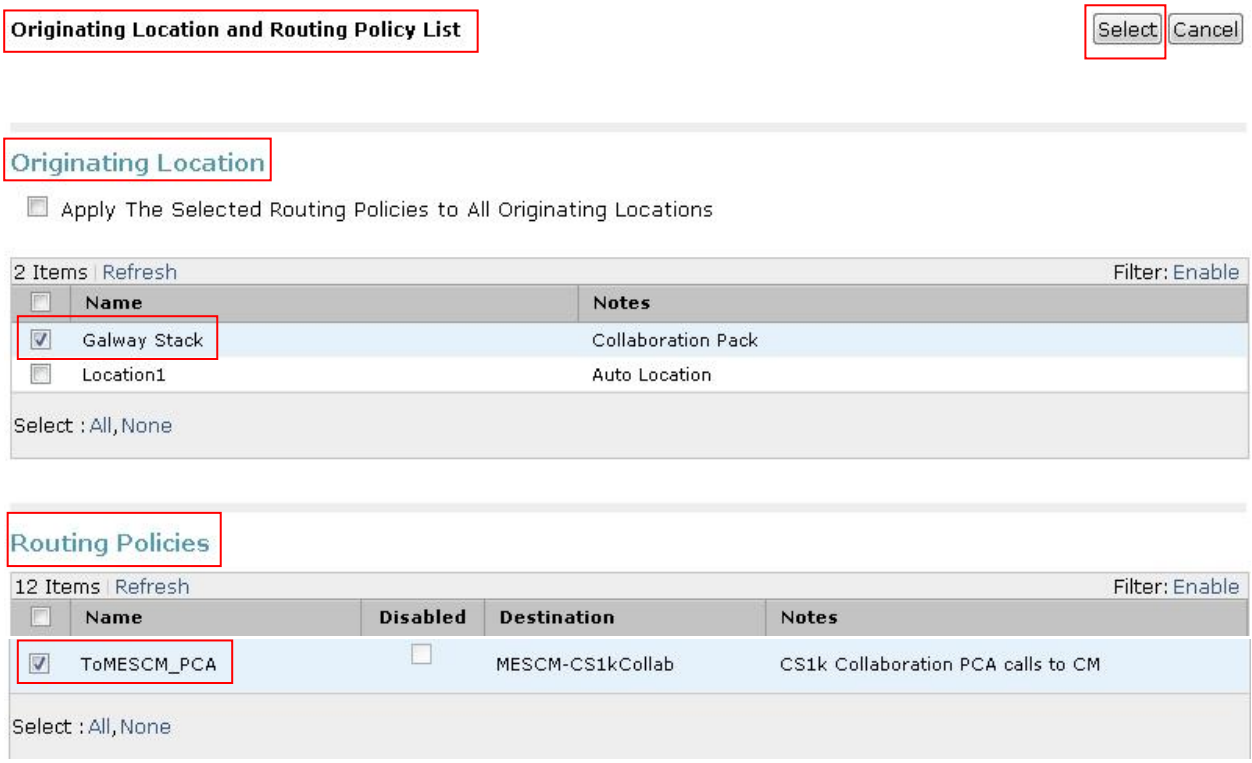
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Configure a dial pattern of 88 to be used to route CS1000 voice calls to Communication Manager as follows. In the **General** section, enter the dial **Pattern** for calls to Communication Manager (e.g. **88**). Because all voice calls to Communication Manager will be 9 digits in length, enter **9** 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**.



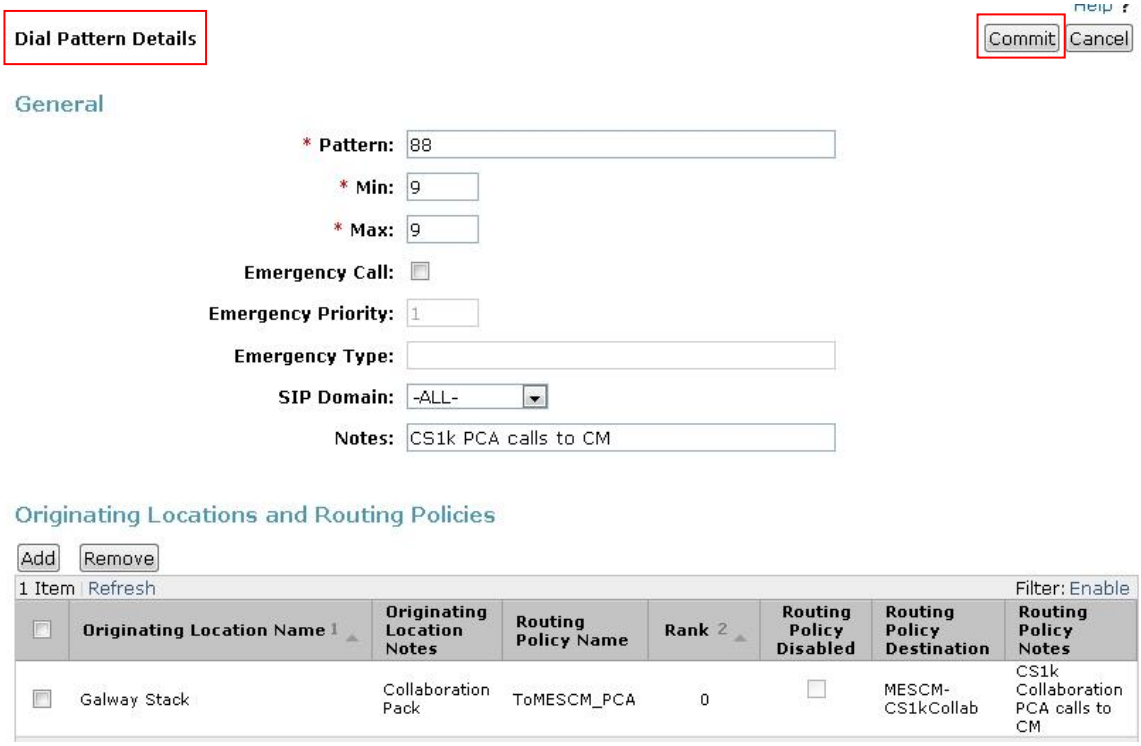
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The **Originating Locations and Routing Policy List** page opens. In the **Originating Locations** table, select **Galway Stack**. In the **Routing Policies** table, select the Routing Policy that should be used to route the digits (e.g. **ToMESCM_PCA**).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



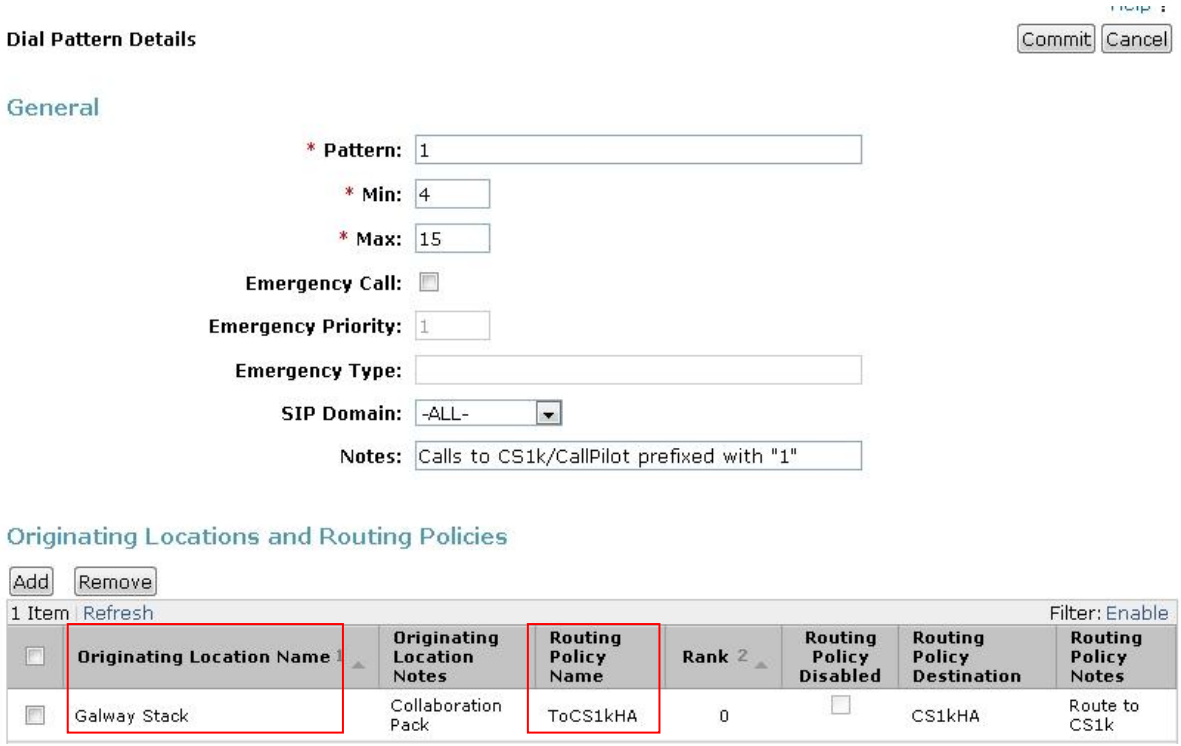
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The **Dial Pattern Details** page is displayed. Click **Commit** to save.



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Since all calls from Collaborated clients bound for the CS1000 will have a leading digit '1' inserted, the dial pattern for those calls is simplified into a single entry. The dial pattern '1' is also used for "drag and drop spotlight calls" from Flare Experience clients, 5-digit station calls on CS1000 and calls to CallPilot voicemail. Follow the same steps described above to configure a dial pattern of 1 to be used to route Communication Manager calls to the CS1000. 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 CS1000 (i.e. **ToCS1kHA**).



Note: For 112 / 911 Emergency calls from Collaboration Pack 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 CS1000 phone which may aid in locating the caller in this case. To do this a second SIP Entity can be created for CS1000 (e.g. called "CS1k_Emergency") with no adaptation applied (therefore no leading digits will be deleted from the origination address as documented in **Section 6.3**). 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 CS1000 for further handling per standard emergency services. Detailed instructions and screenshots of this configuration are not shown in this document.

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7. Configure Avaya Aura® Communication Manager

This section describes the steps required to configure Communication Manager (as an Evolution Server) to support Converged endpoints for CS1000 with Avaya Aura® Midsize Enterprise. These instructions assume the Avaya G430 (or equivalent) Gateway is already configured on Communication Manager.

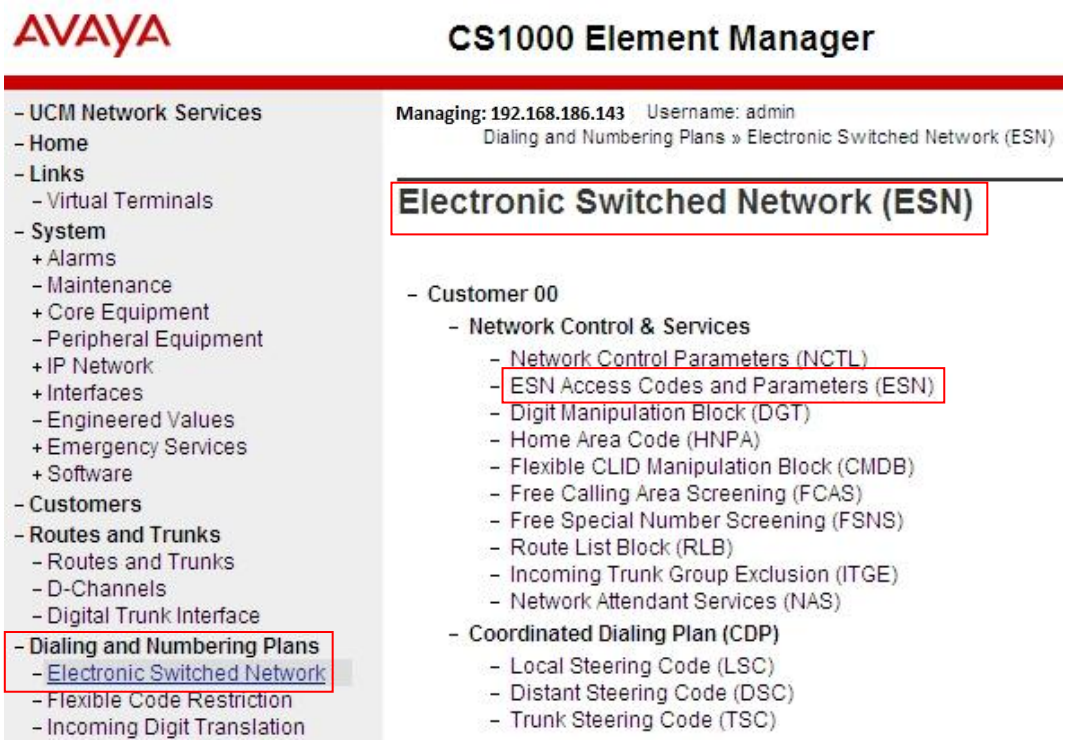
The following administration steps will be described:

- 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

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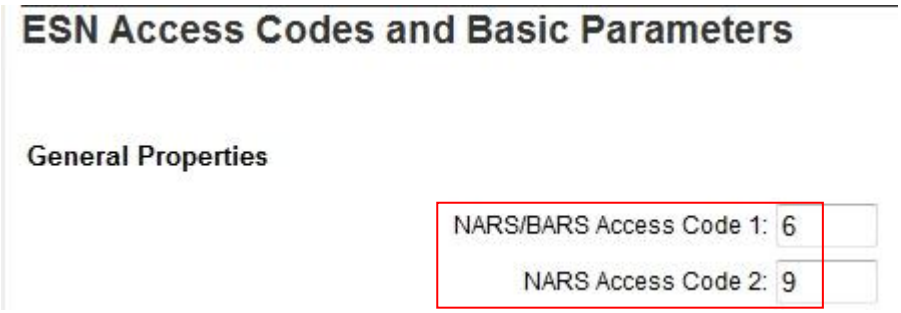
7.1. Verify System Access Codes match

To allow users to utilize Converged dialing plans to route calls, verify Communication Manager AAR and ARS access codes match CS1000 Access Codes. From System Manager Home Page under the Services category, navigate to UCM Services for CS1000 management (as described earlier in **Section 5.2**). 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 CS1000 in the Element Type column (e.g. EM on cs1kcores1). From the **CS1000 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)**.



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The **ESN Access Codes and Basic Parameters** page appears. Take note of the numbers used for CS1000 **NARS/BARS Access Code 1** (e.g. **6**) and **NARS Access Code 2** (e.g. **9**).

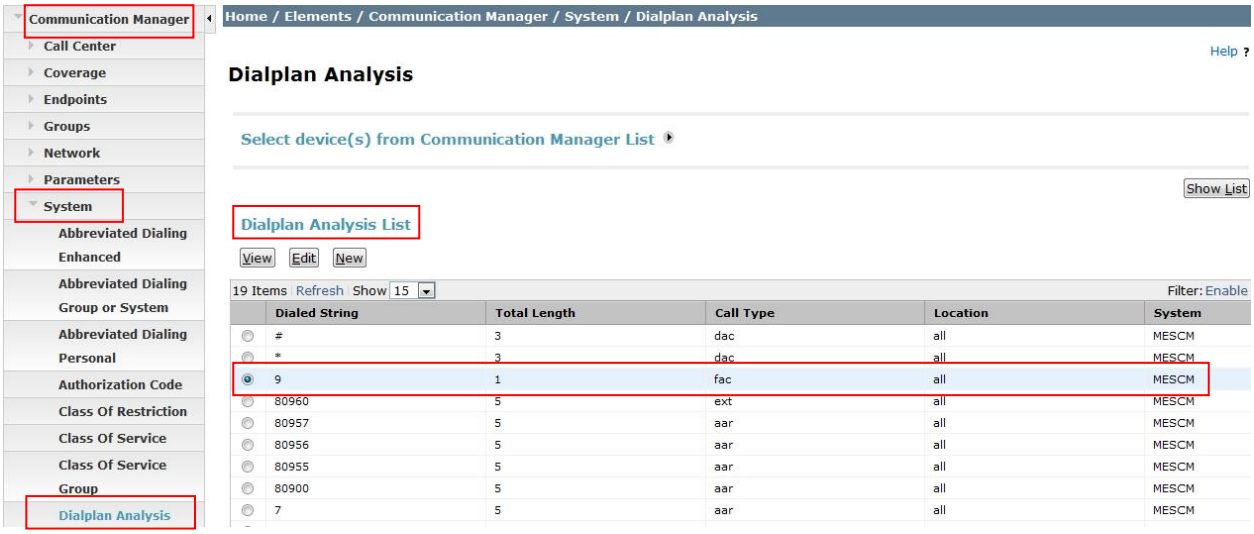


From the main System Manager 6.2 page under **Elements**, click on **Communication Manager**.



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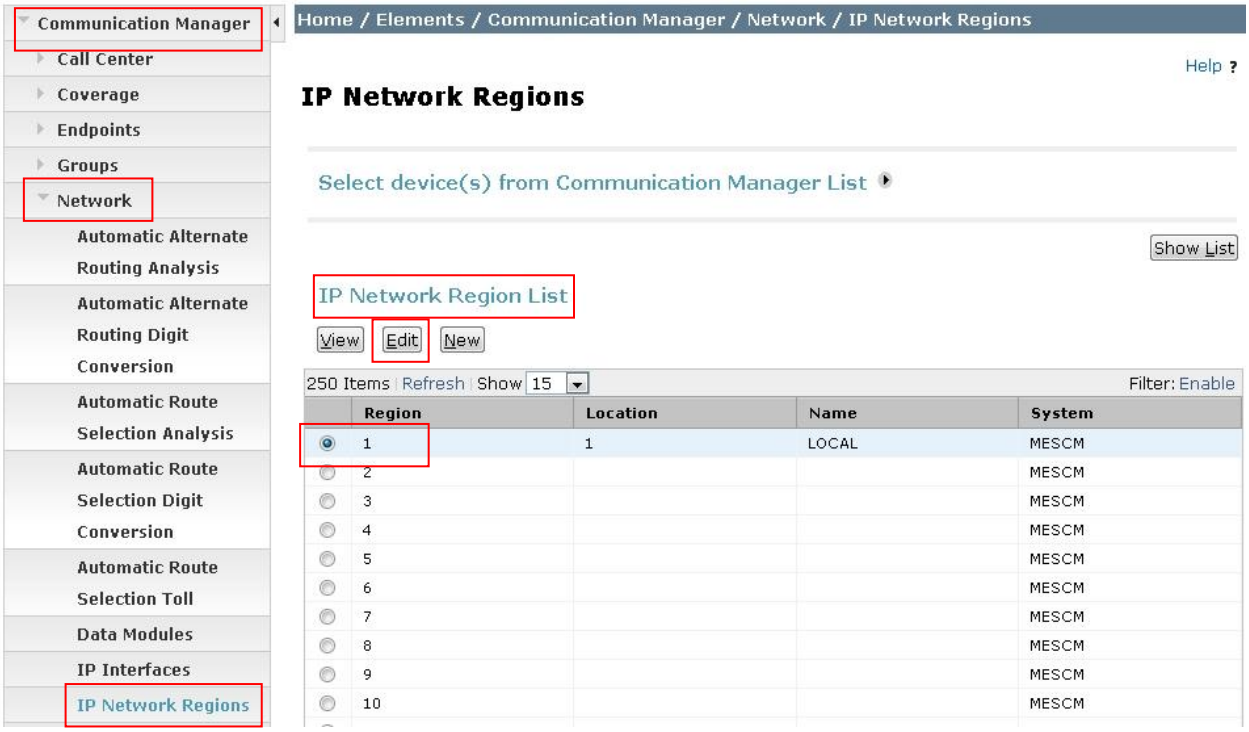
Under the **Communication Manager** list on the left hand side, expand the **System** list and select **Dialplan Analysis**. It is presumed that in the initial configuration of the Midsize Enterprise / Communication Manager, dial codes of 9 and 6 may already have been configured as they are commonly 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).



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7.2. Verify IP Network Region – SIP Domain

From the main System Manager 6.2 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.

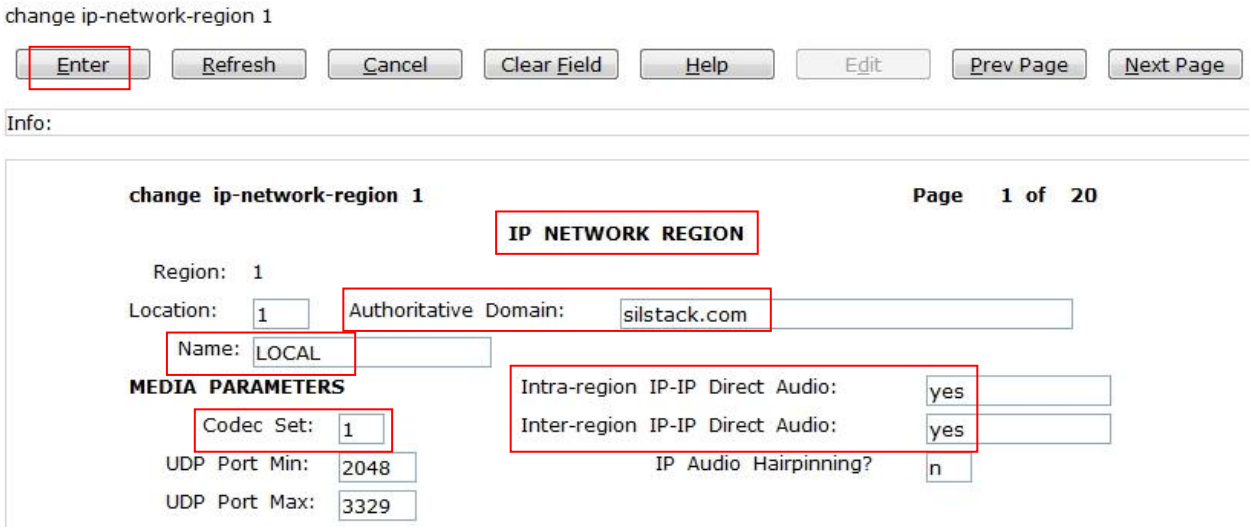


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Page 1 of the **IP NETWORK REGION** appears. Verify or enter the following values and use default values for remaining fields.

- **Authoritative Domain:** Enter the correct SIP domain for the configuration (e.g. **silstack.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**

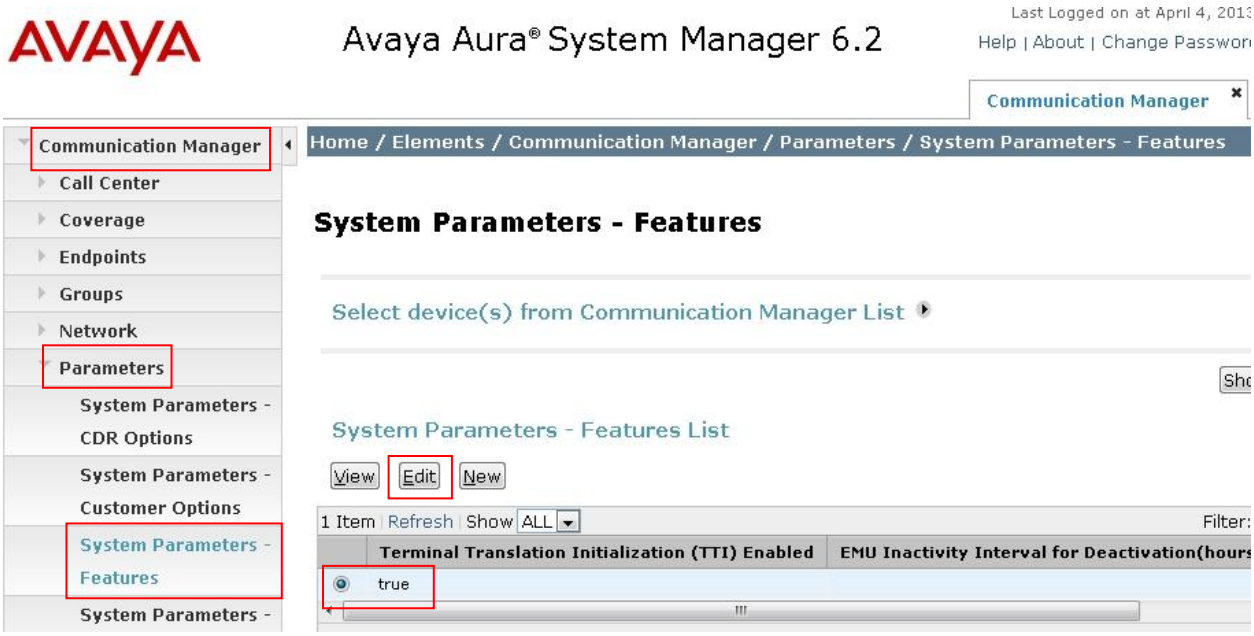
Click on **Enter** if any changes were made otherwise click on **Cancel**.



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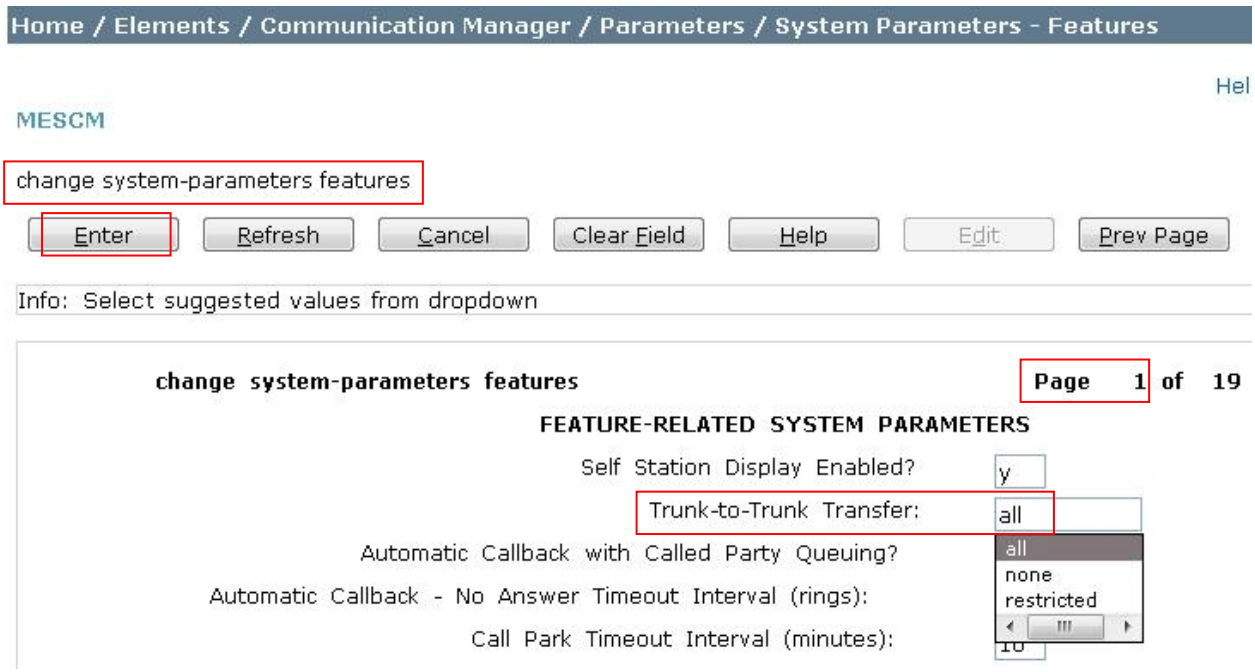
7.3. Configure Trunk-to-Trunk Transfers

From the main System Manager 6.2 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 **true)**, and click on **Edit**.



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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.

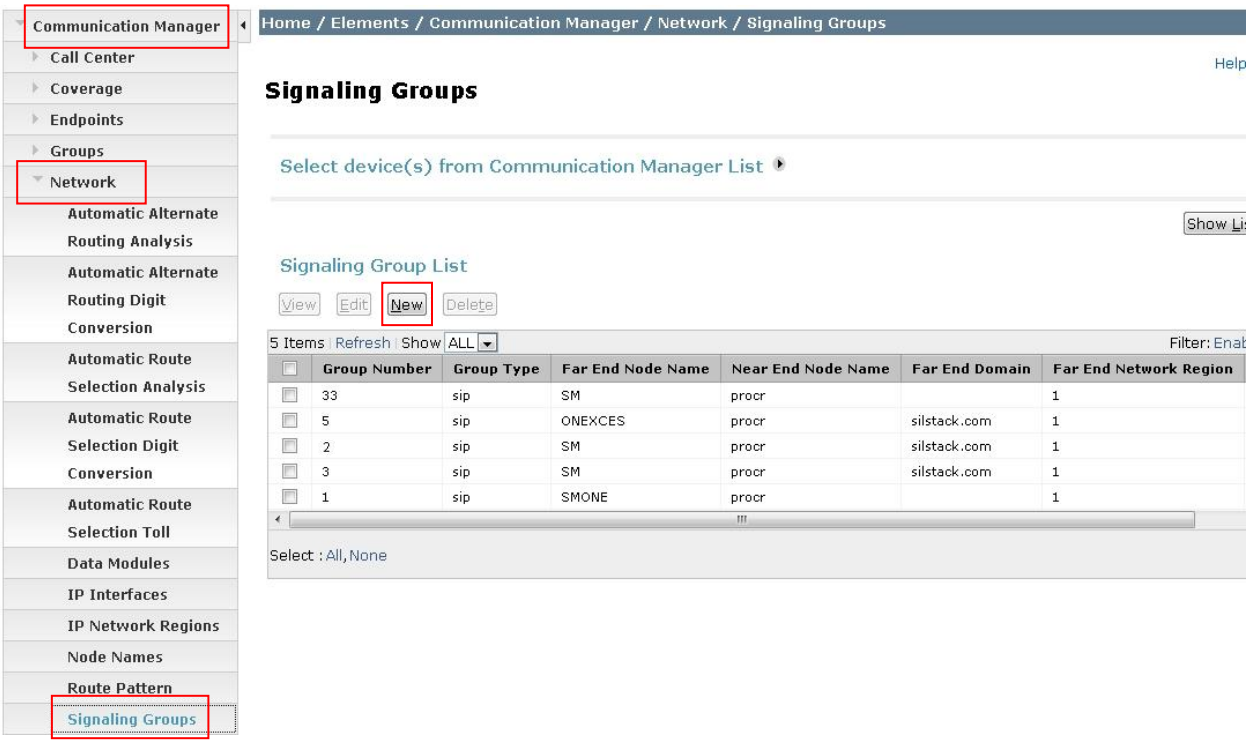


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7.4. Administer SIP Signaling Group

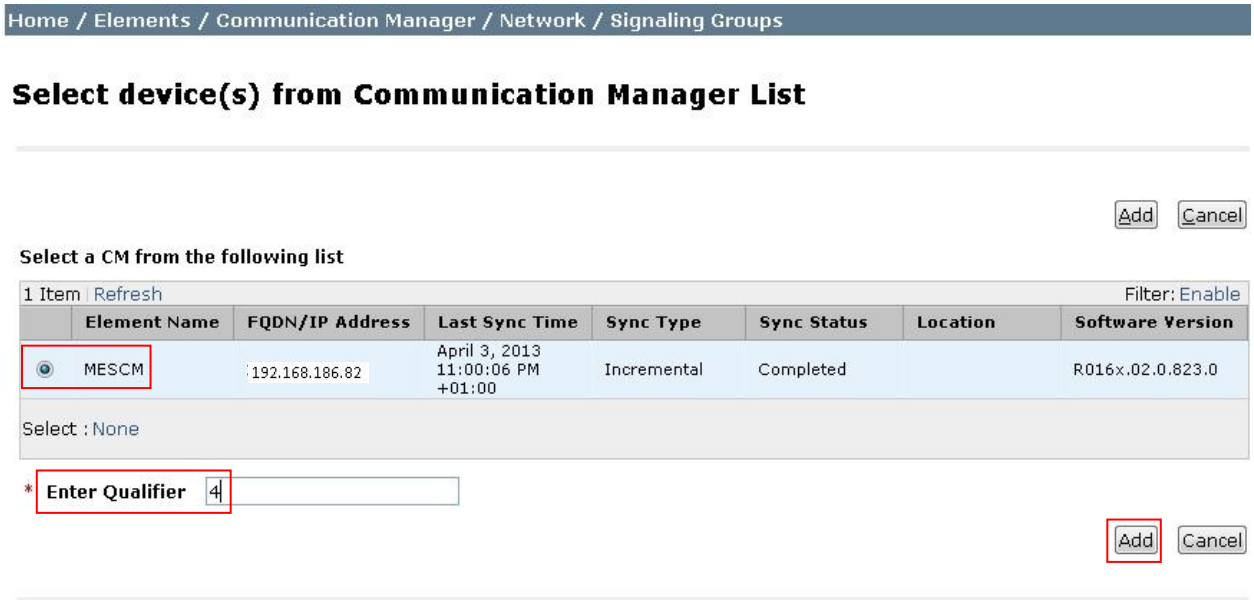
Signaling Group 3 and an associated SIP Trunk Group 3 is presumed to have already been setup and configured for the solution with the ME installation. Signaling Group 4 and an associated SIP Trunk Group 4 will be used for all Communication Manager traffic to the CS1000 and needs to be configured to use a different TLS port from the default.

From the main System Manager 6.2 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**.



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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**.

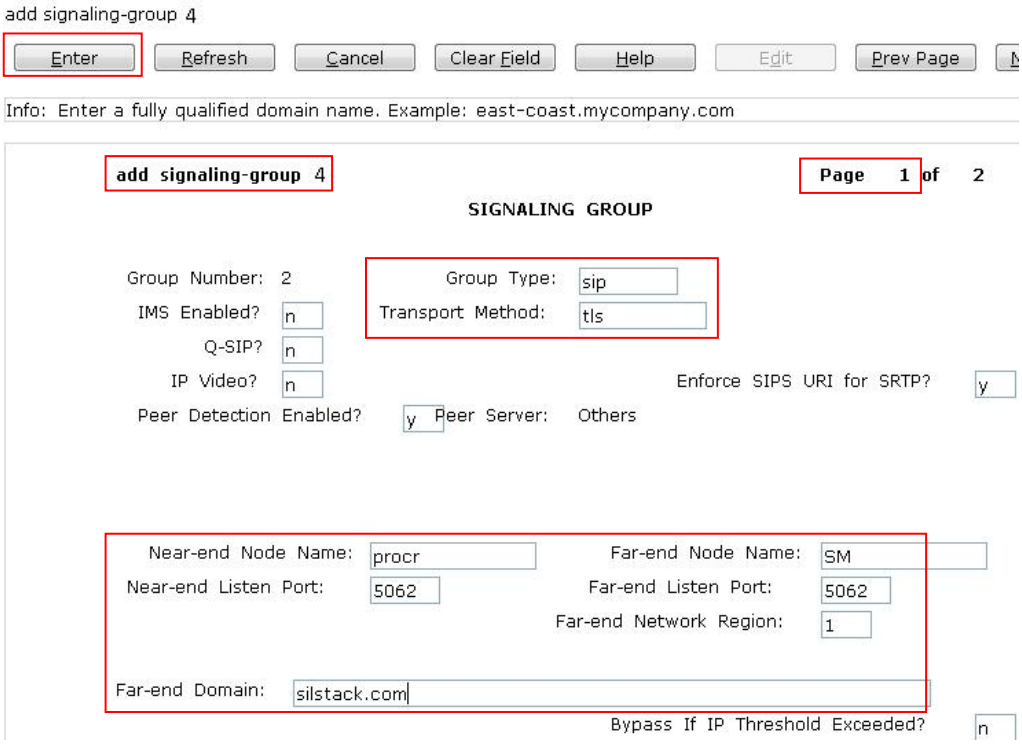


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Page 1 of the **add signaling-group 4** page appears. Select or enter the following values and leave all other fields as default.

- **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. **silstack.com**).

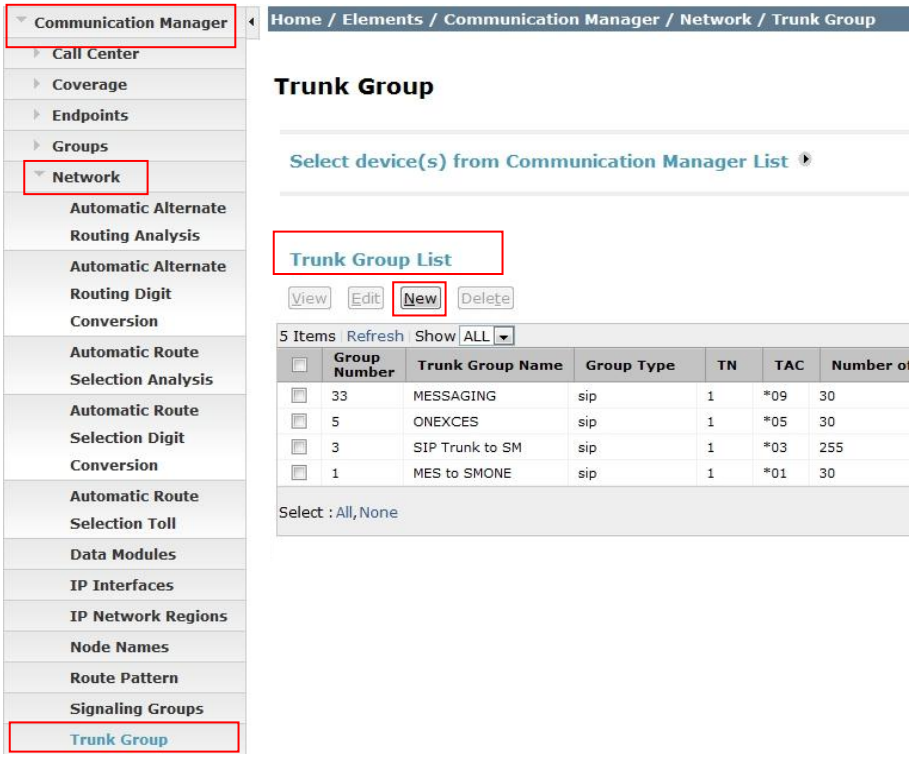
Click on **Enter** to save the new signaling group.



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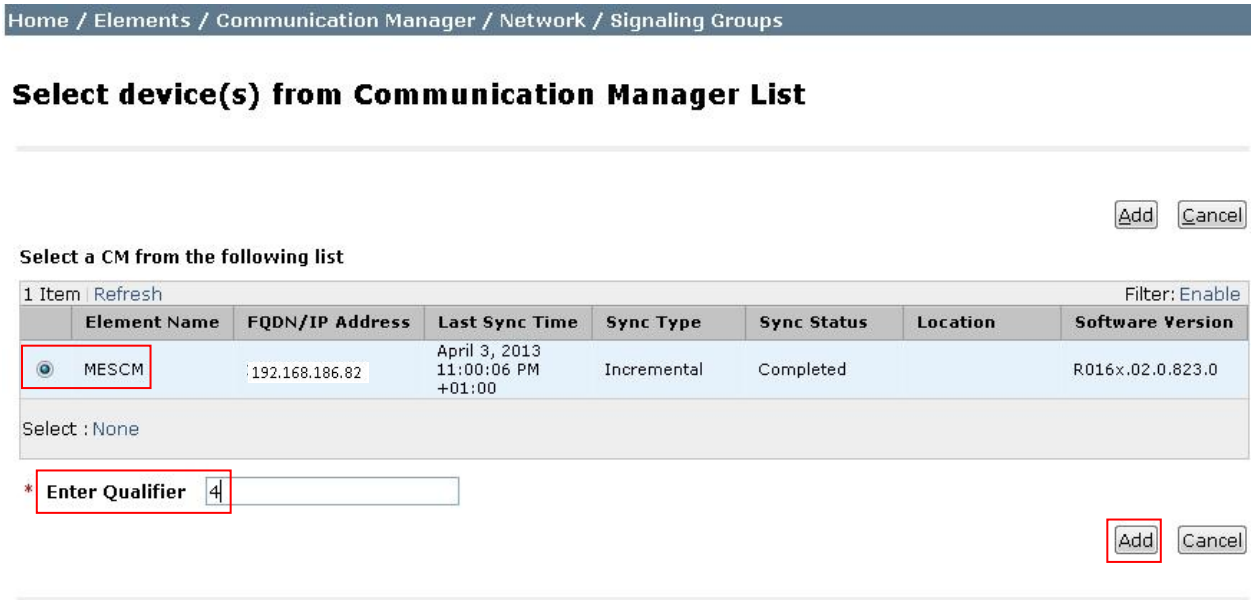
7.5. Administer SIP Trunk Group

Trunk Group 4 will use signaling group 4 created in **Section 7.4** and will be used for all traffic to the CS1000. From the main System Manager 6.2 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**.



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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**.

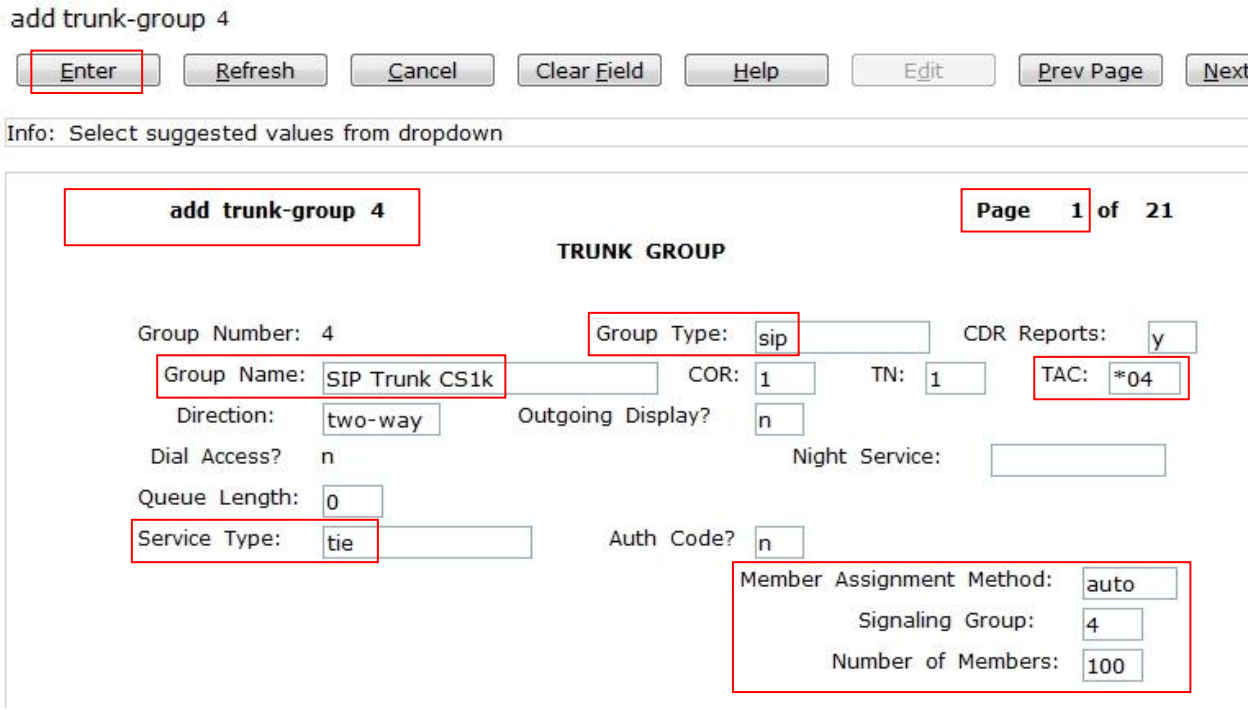


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Page 1 of the **add trunk-group 4** page appears. Select or enter the following values and leave all other fields as default.

- **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 7.4**.
- **Number of Members:** Enter **100**.

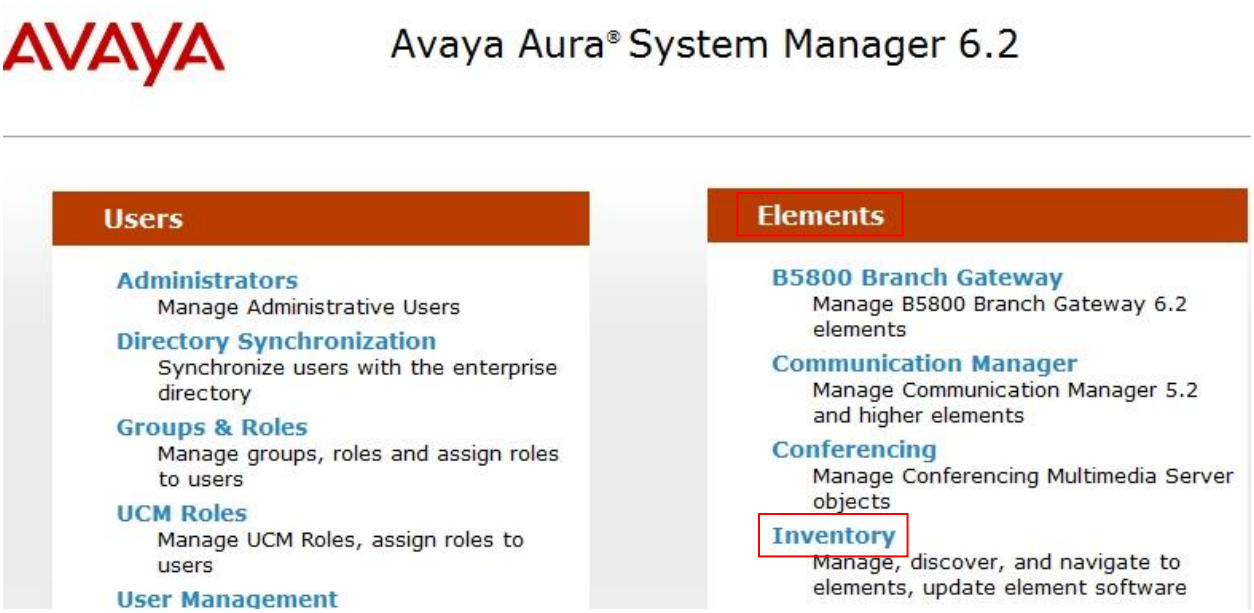
Click on **Enter** to save the new signaling group



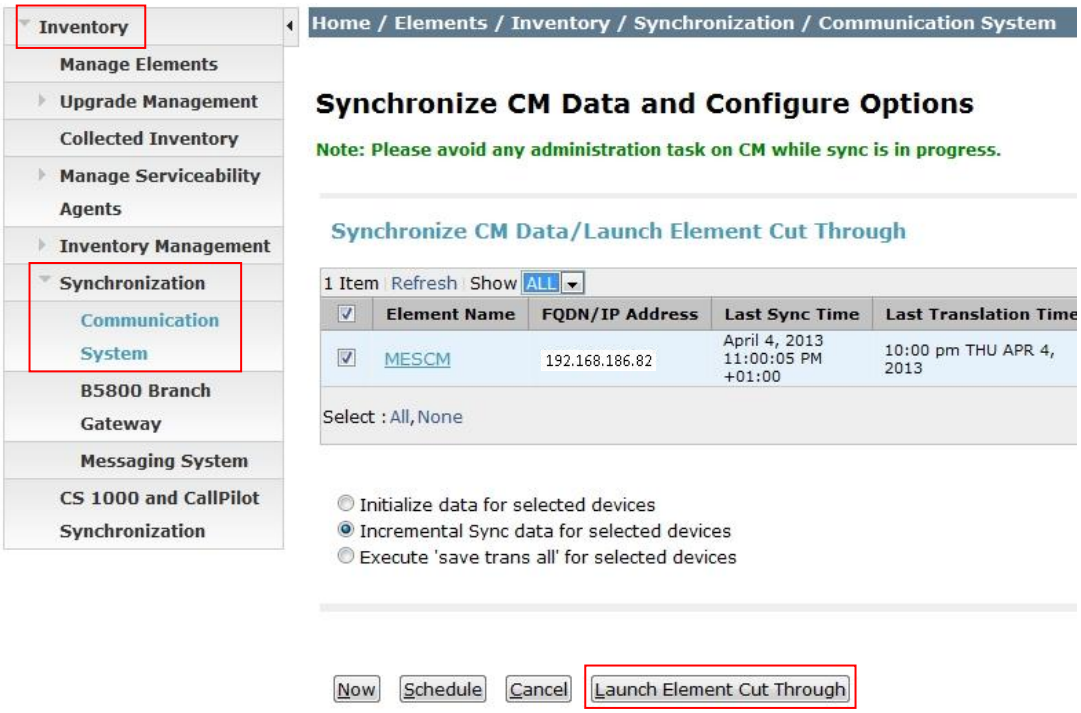
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7.6. Verify Signaling Group and Trunk Group are in-service

After Signaling Group 4 and Trunk Group 4 have been added, ensure that both are up and in-service. From the main System Manager 6.2 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**.



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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).

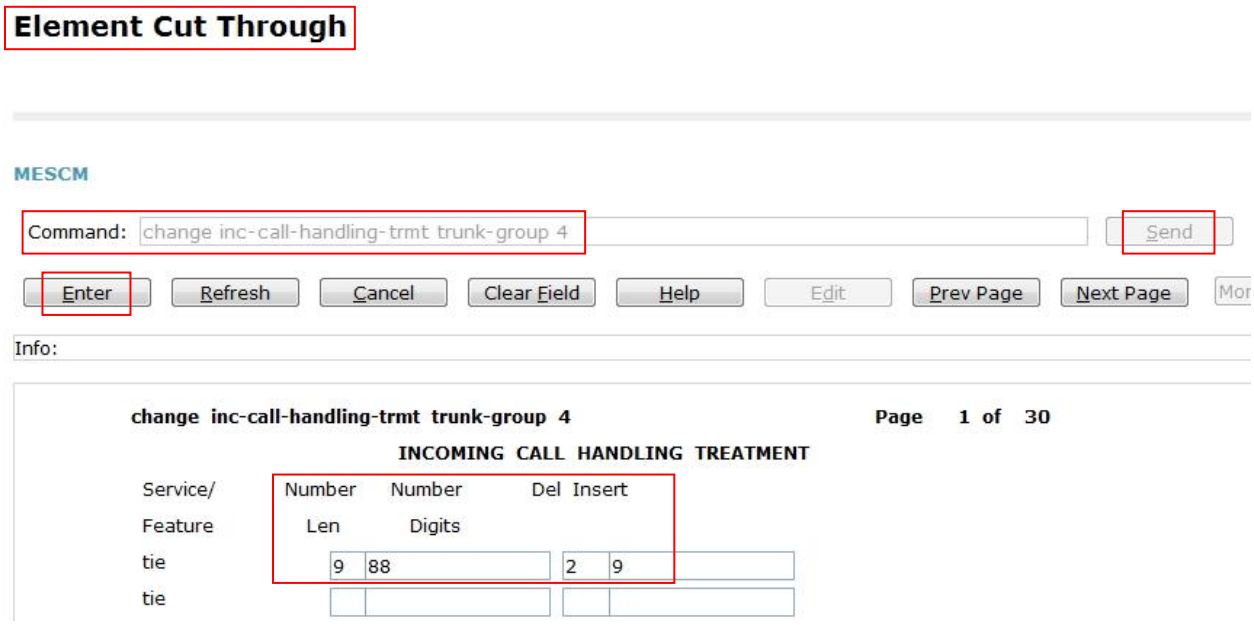


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7.7. Configure Incoming Call Handling for Trunk Group

In order to strip the routing digits of 88 for PCA calls from the CS1000 to Collaboration clients, an Incoming Call Handling treatment rule is needed on trunk group 4 in Communication Manager. This rule will delete the prefix 88 digits and insert the digit 9 instead. This will then route the call to ARS where ARS digit conversion will keep it local rather than looping it back to the CS1000.

Using the same **Element Cut Through** window used in the previous **Section 7.6**, in the **Command** box, enter **change inc-call-handling-trmt trunk-group 4** and click on **Send**. The INCOMING CALL HANDLING TREATMENT page appears. In the **Number Len** box, enter **9** (this is based on the prefix 88 + the 7-digit collaborated endpoint DNs). In the **Number Digits** box, enter **88** to trigger the number match. In the **Del** box, enter **2** to delete the two 88 digits. In the **Insert** box, enter **9**. Click on **Enter** to accept the change.



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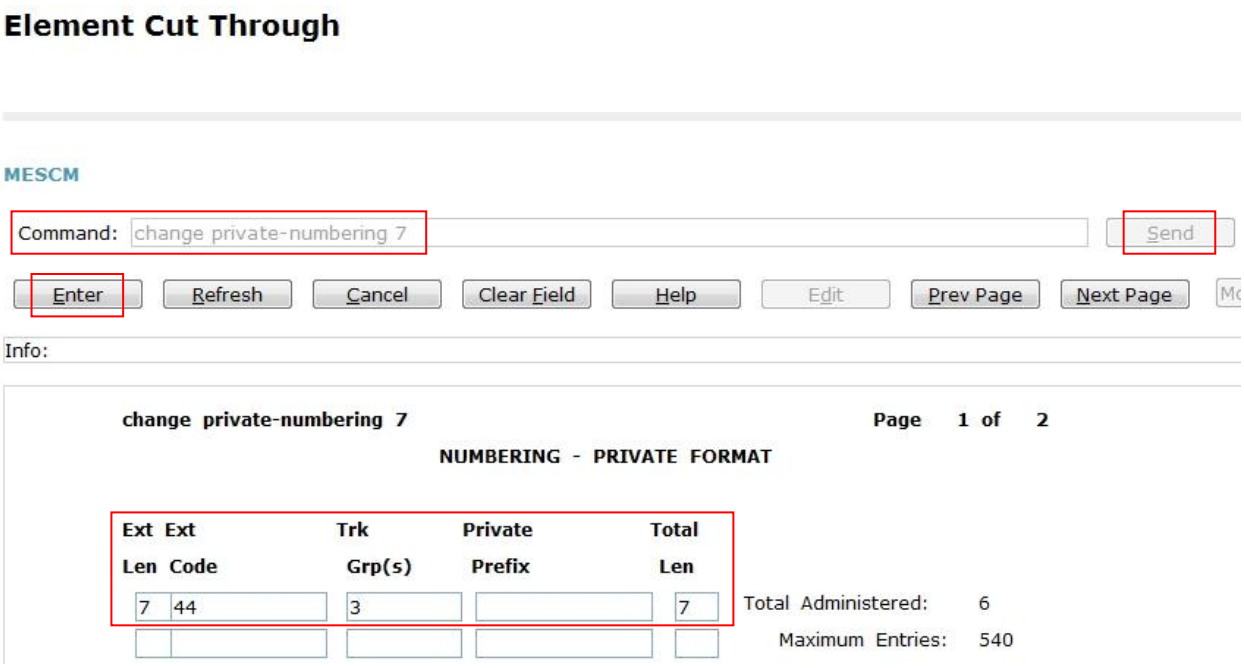
7.8. Administer Private Numbering Plan

The full extension numbers used for the collaboration 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 7.7**, 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 SIP endpoints).
- **Trk Grp(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.



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7.9. Administer Public Numbering Plan

The full extension numbers used for the collaboration 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 7.8**, 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).
- **Trk Grp(s):** Enter **4** as the trunk group in this case.
- **Total Length:** Enter **5** since a private prefix was not defined.

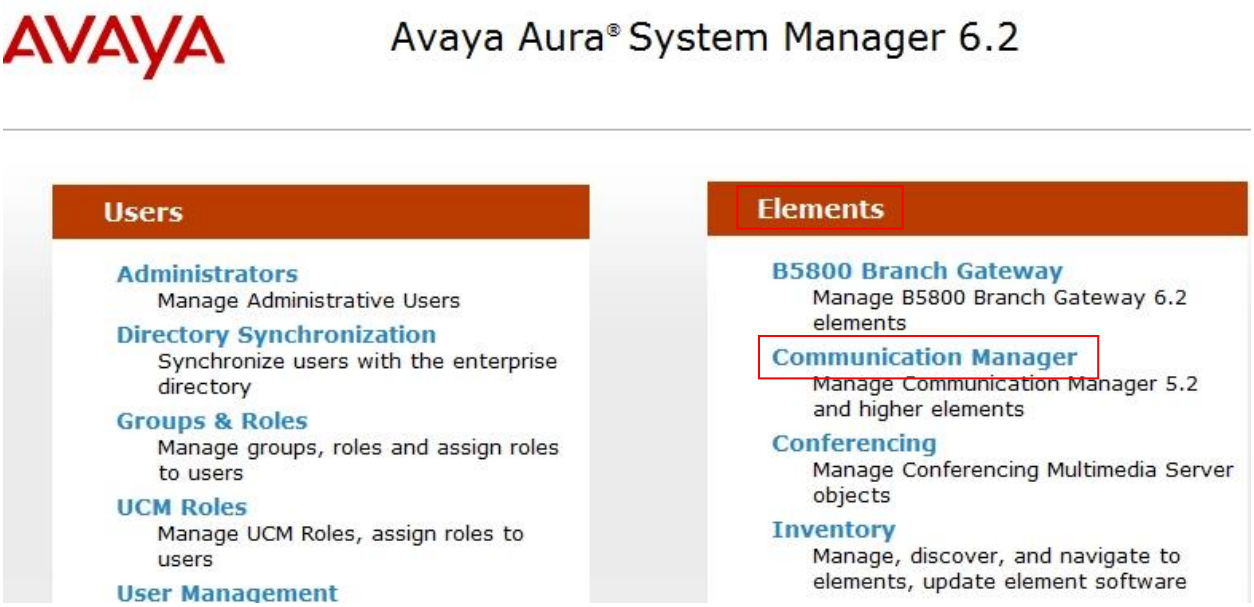
Click on **Enter** to accept the change.



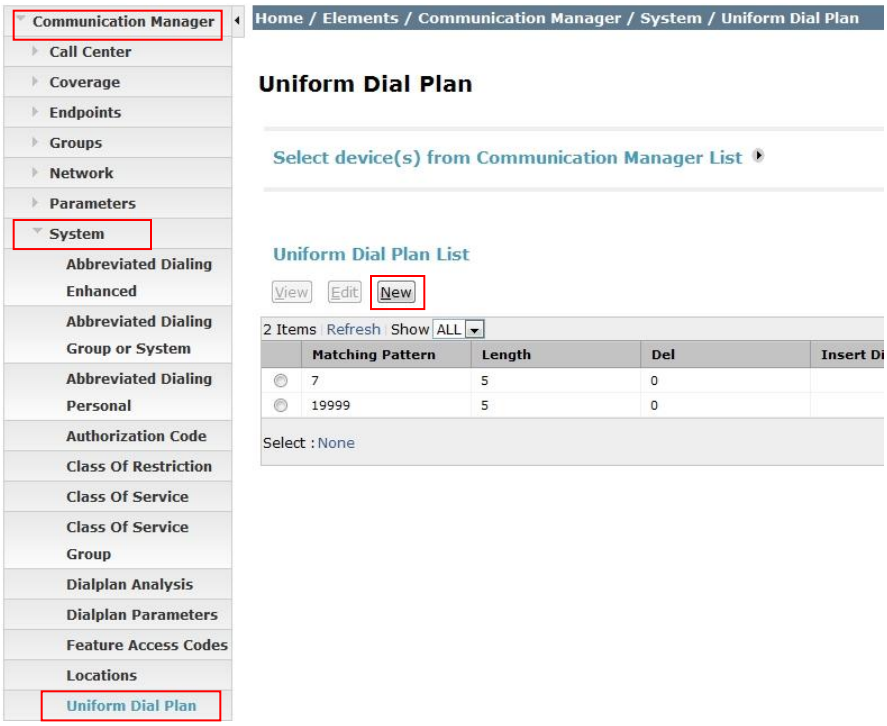
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7.10. Administer Uniform Dial Plan

From the main System Manager 6.2 page under **Elements**, click on **Communication Manager**.

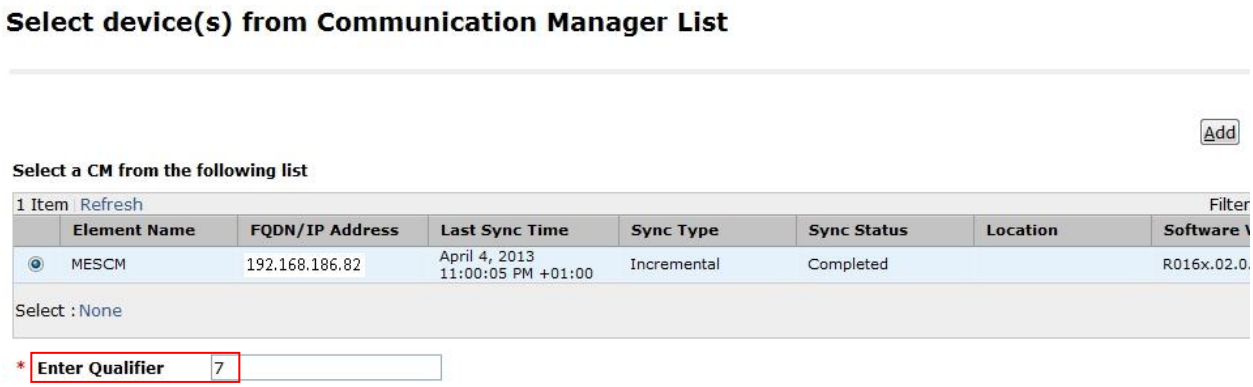


Under the **Communication Manager** list on the left hand side, expand the **System** list and select **Uniform Dial Plan**. Click on **New**.



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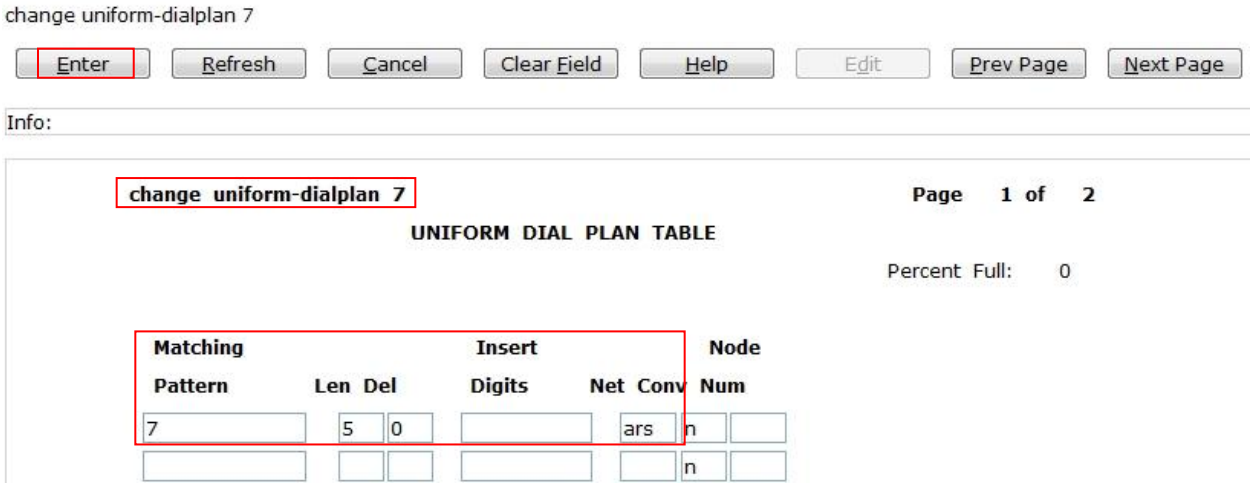
As all extensions on the CS1000 in the sample configuration begin with the digit 7, enter **7** into the **Enter Qualifier** box.



The **change uniform-dialplan 7** window appears. In the reference configuration, **7xxxx** was used as the uniform dialing plan for Converged Users dialing across CS1000 and the Collaboration Pack 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.

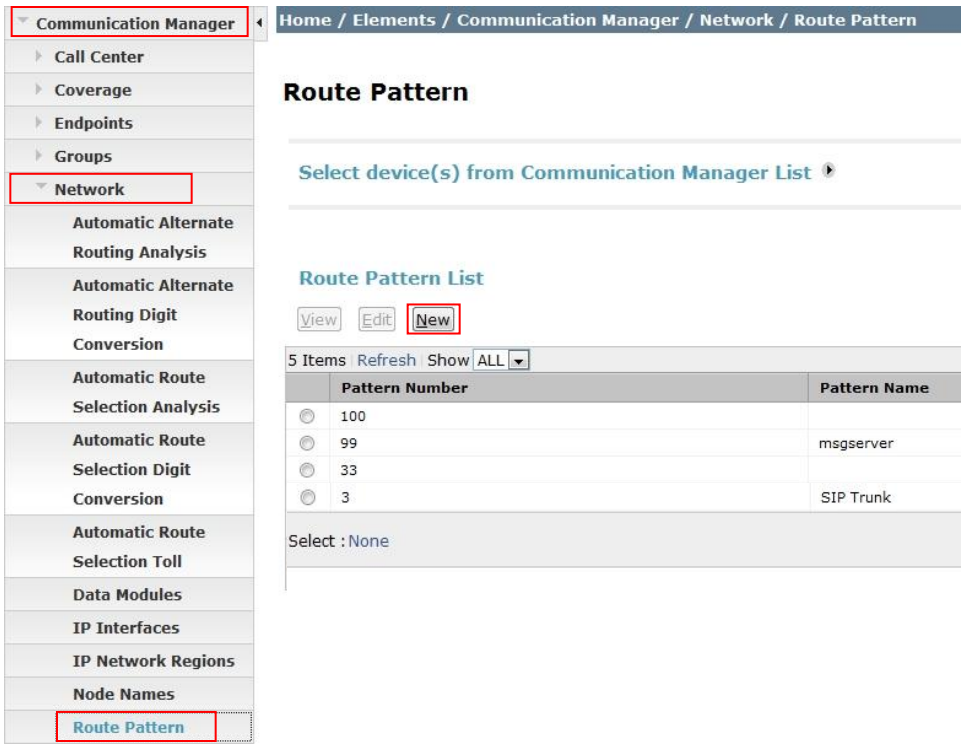


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7.11. Administer Route Pattern

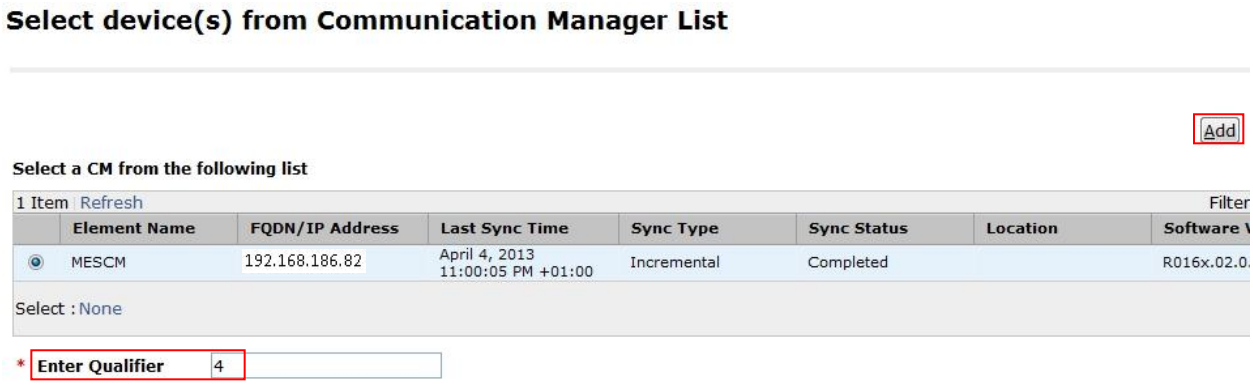
This section describes the configuration of the Route Pattern used in Communication Manager for the routing of calls to CS1000. All calls from ARS will use this route pattern 4. Route Pattern 3 is created during the ME server installation and is dedicated for the IMS signaling required by the Collaboration 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 Pattern 3.

From the main System Manager 6.2 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**.



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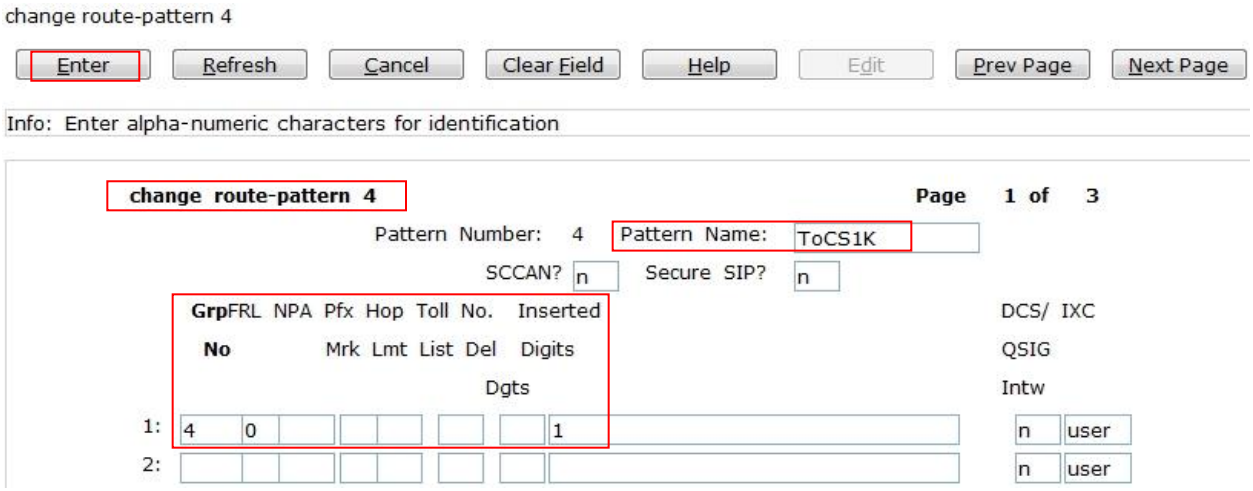
Enter the route pattern number **4** into the **Enter Qualifier** box and click on **Add**.



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

- **Pattern Name** 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 CS1000.

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

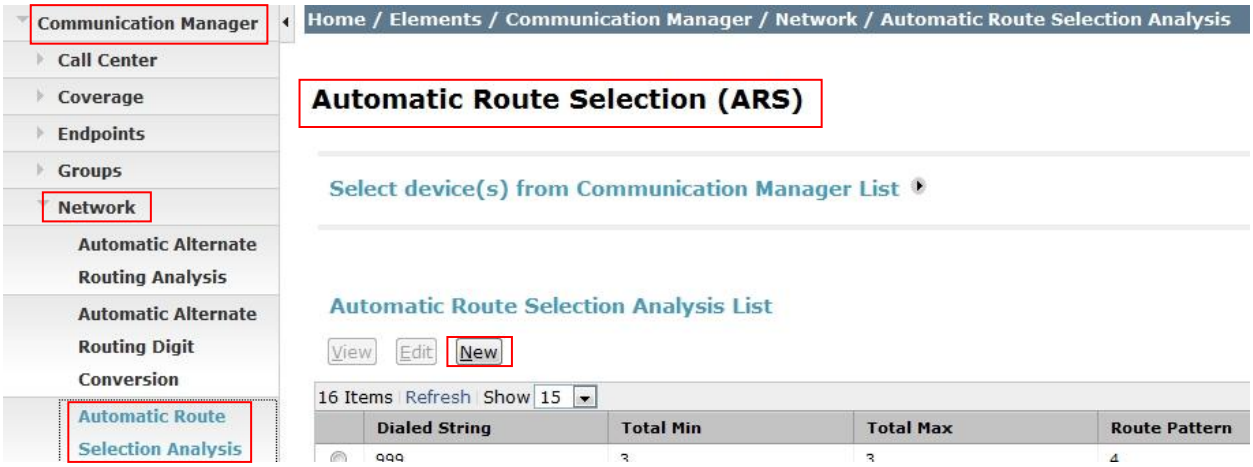


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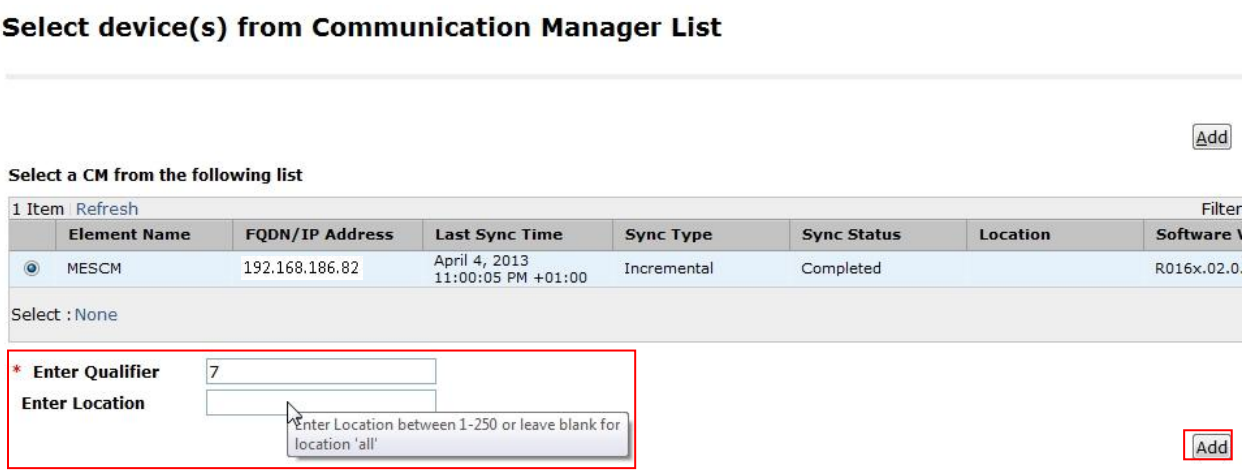
7.12. Administer ARS Analysis

This section details the configuration of the ARS pattern used in the reference configuration for routing calls between collaboration clients and CS1000 stations. All dialed numbers identified on the ARS table will be sent to Session Manager and routed to the CS1000 via Route pattern 4 created in **section 7.11**.

From the main System Manager 6.2 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 CS1000 start with digit 7 while all stations on Collaboration Pack 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**.

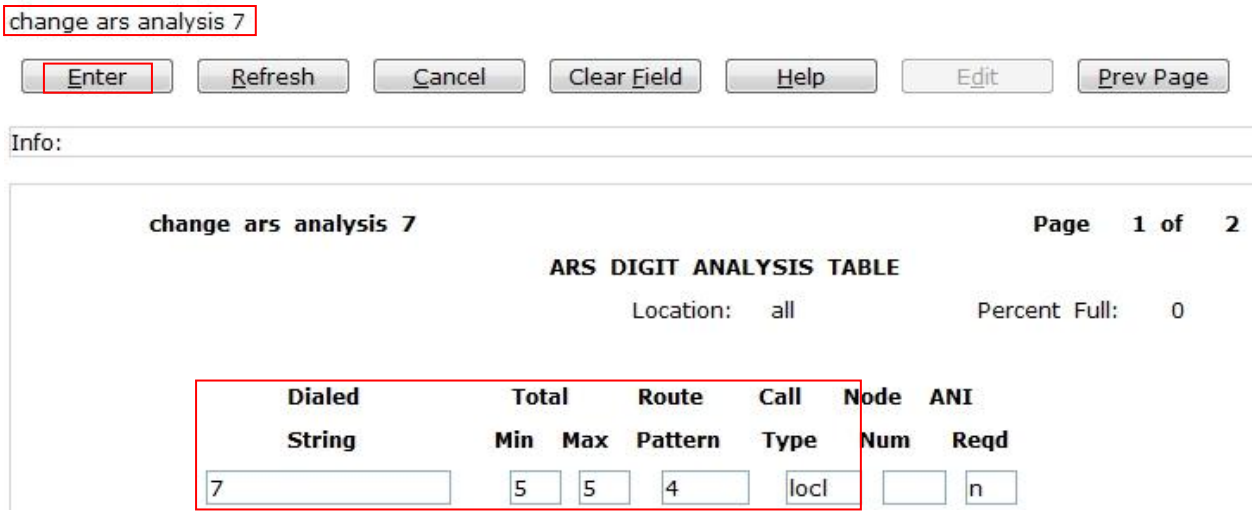


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The **change ars analysis 7** page appears. Enter the following values per the sample configuration example:

- **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.



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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.

display ars analysis 44

Enter | Refresh | Cancel | Clear Field | Help | Edit | Prev Page

Info:

display ars analysis 44 **Page 1 of 2**

ARS DIGIT ANALYSIS TABLE

Location: all Percent Full: 0

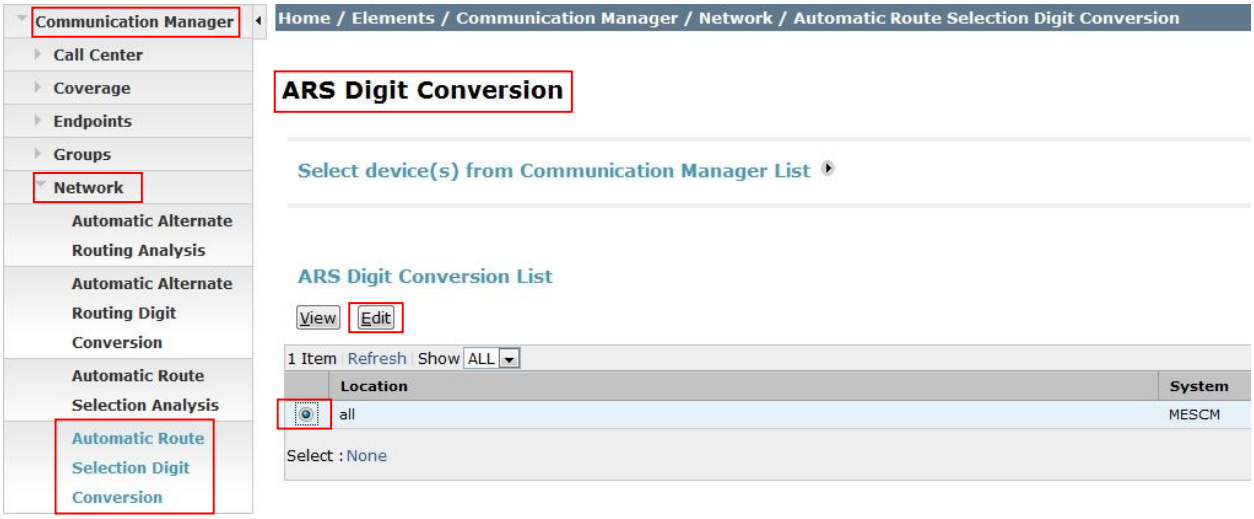
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
44	7	7	4	locl		n
5	5	6	1	locl		n
6	5	6	1	locl		n
7	5	5	4	locl		n

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7.13. Administer ARS Digit Conversion

This section details the configuration of the ARS Digit-Conversion rule used in the sample configuration for keeping inbound 7-digit calls local rather than routing back out to the CS1000. In order to support drag/drop calling from the Flare Experience client contact fan (which is a 7-digit call) to other Converged Users, changes were made to the dial plan parameters form. The changes made to the Dial Plan Parameters form routed 7-digit traffic to the 44xxxxx range off-net so that a 7-digit call will ring the CS1000 station first (invoking PCA/SIMRING capability) rather than just calling the Collaboration Client directly. Without a rule to keep inbound 7-digit traffic local, PCA calls to 7-digit SIP clients would also be routed off-net creating a loop condition. Inbound 7-digit calls will be kept local by using an ARS Digit-Conversion rule that states that the 44xxxxx range represents extensions. The Incoming-Call Handling rule created in **Section 7.7** inserts the digit 9 which routes the call to ARS. Before ARS Analysis is invoked, ARS Digit-Conversion rules are applied.

From the main System Manager 6.2 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 Digit Conversion**. The **ARS Digit conversion** page appears. Click on the radio button to select the device from the Communication Manager List and click on **Edit**.

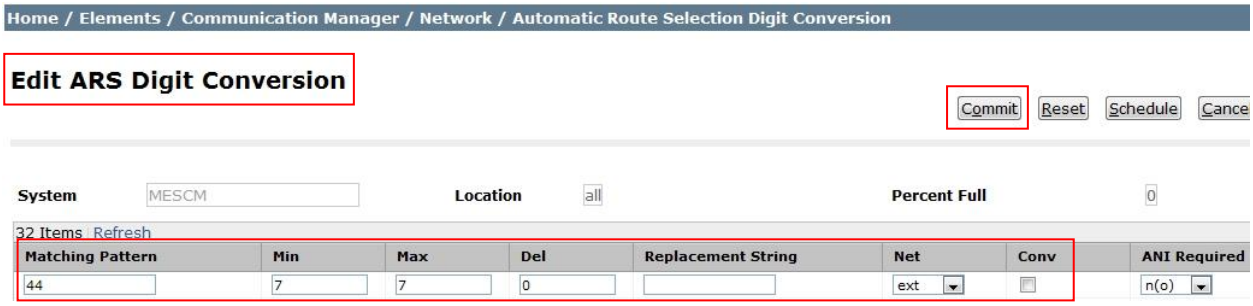


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The **Edit ARS Digit Conversion** page appears. Enter the following values as in the sample configuration example.

- **Matching Pattern** Enter **44** (these are the leading digits for Collaboration client DNs).
- **Min** Enter **7** (length of the Collaboration client extension number).
- **Max** Enter **7** (length of the Collaboration client extension number).
- **Del** Enter **0** (no changes made to the digit string).
- **Net** Enter **ext** from the drop-down list to define this range as an extension to prevent a loop.
- **Conv** Leave this box blank so that no other conversion should be applied to this digit string.

Click on **Commit** to save the data.



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8. User Management

This section describes the details for configuring Converged and Native users across the CS1000 and the Collaboration Pack 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 CS1000 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 CS1000 is registered as a member of the System Manager Security framework.

In the sample configuration, a user with CS1000 DN of 70408 already exists and a Flare Experience 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 CS1000 deskphone but will have a SIP endpoint on the Collaboration Pack, a PCA configuration on CS1000 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 CS1000 for this purpose.

The following administration steps will be described:

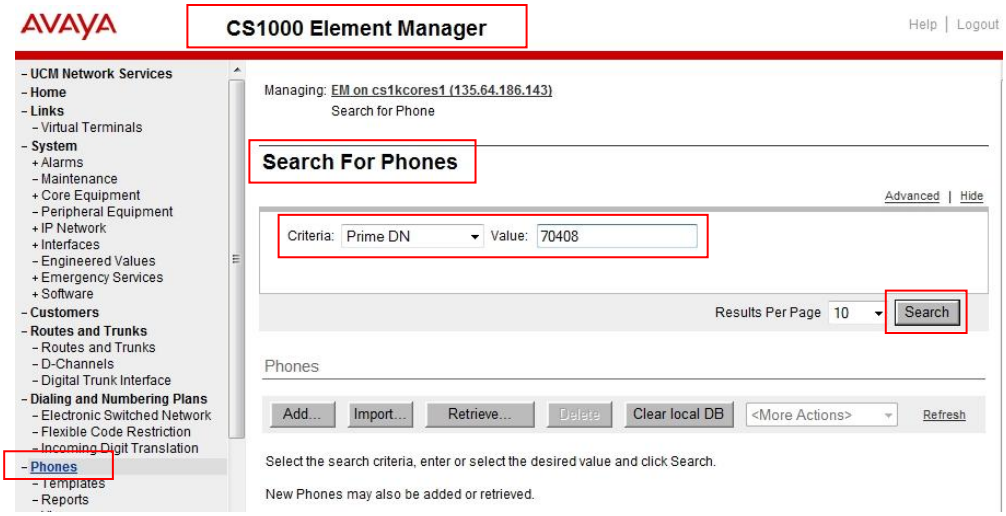
- Confirm existing users in Avaya Communication Server 1000E Element Manager,
- Create User Identities and Communication profiles,
- Personal Call Assistant Configuration (PCA),
- Manual Configuration of Avaya SIP Clients.
- Synchronize CS1000E Profile to User Identities in System Manager,
- Add PCA to CS1000 User Communication profiles.

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

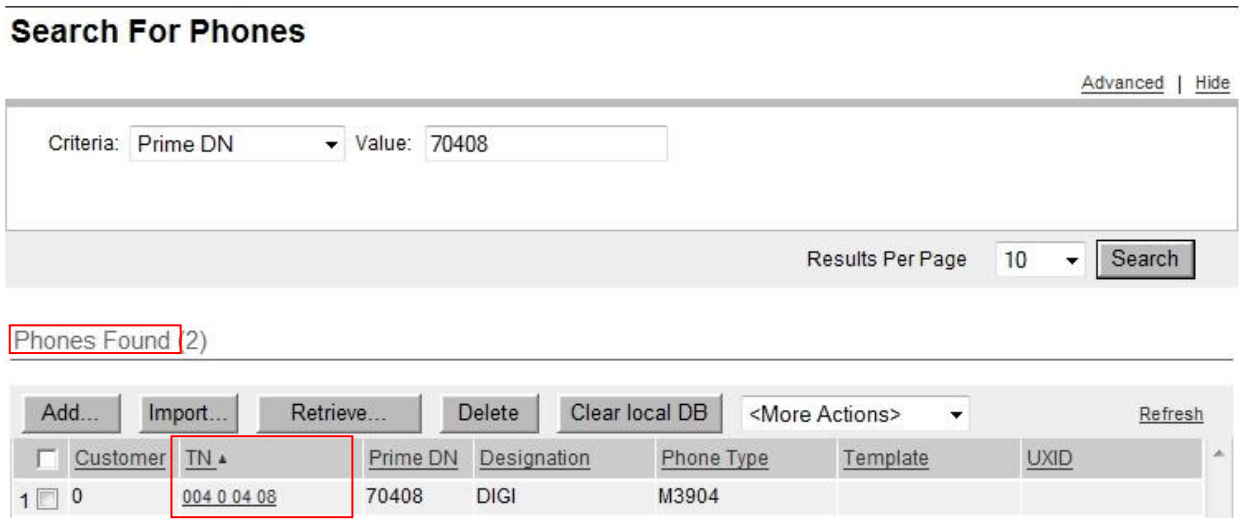
It is presumed that existing CS1000 users have been previously created using CS1000 Element Manager which has configured the main endpoint phone Terminal Number (TN) / Directory Number (DN) and Call Party Name Display (CPND - First Name, Last Name). For each CS1000 user defined in Element Manager a corresponding user identity must be added in System Manager. The **First Name** and **Last Name** of the user must match exactly on both CS1000 Element Manager and System Manager User Management. This is important for proper Presence synchronization and import synchronization for user CS1000 and CallPilot endpoint profiles.

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Using the UCM Services link in the System Manager main page, access the **CS1000 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 CS1000 user (e.g. **70408**). Click on **Search** button.



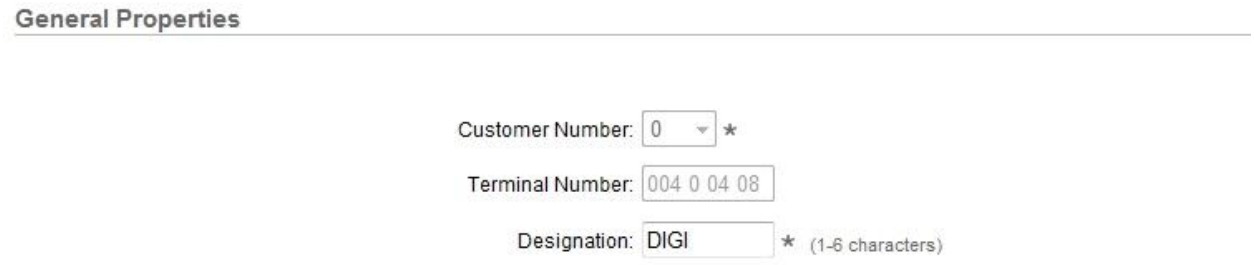
The result of the search comes back and displays **Phones Found**. Click on the TN hyperlink for the phone entry found (e.g. **004 0 04 08**).

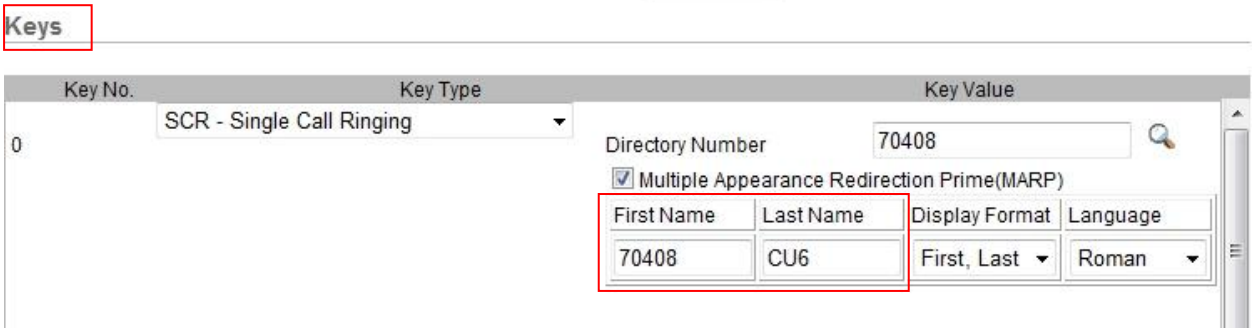


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The **Phone Details** page now appears. Scroll down to the **Keys** properties section and make note of the **First Name** and **Last Name** of the user (e.g. **70408**, **CU6**).







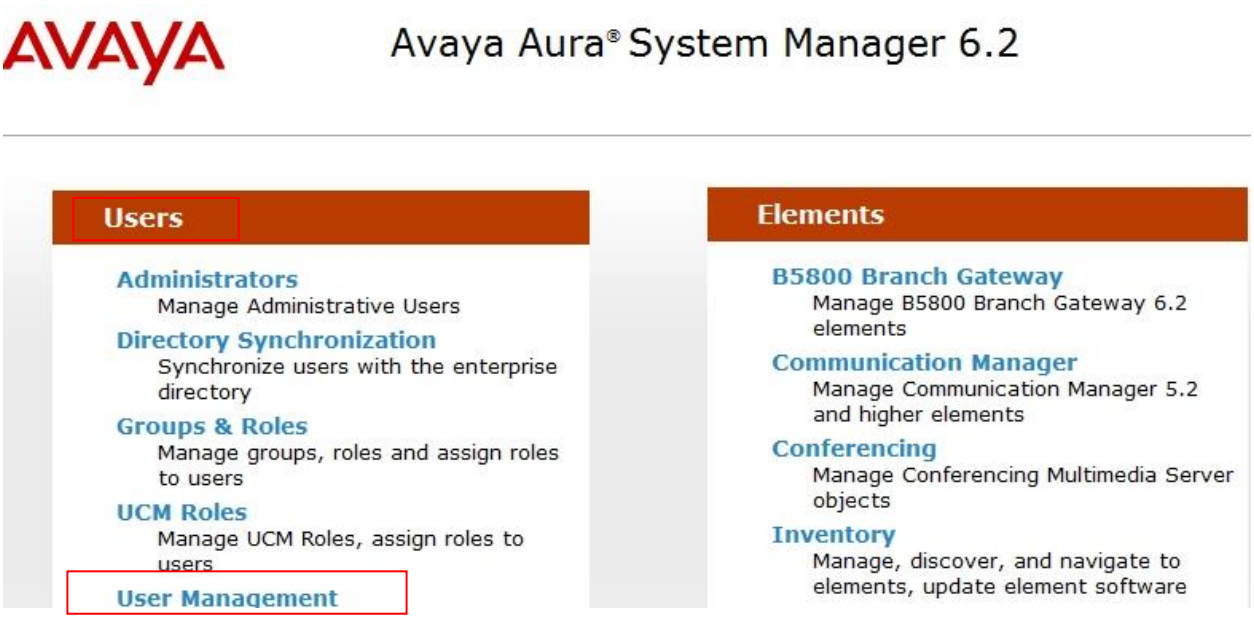
Ensure Feature FTTC is configured as "Unrestricted Conf. or Transfer" to ensure conferences and transfers work across SIP trunk between CS1000 and Collaboration Pack as shown below.



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8.2. Create User Identities and Communication profiles

To create new users on System Manager, go to the main System Manager 6.2 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.

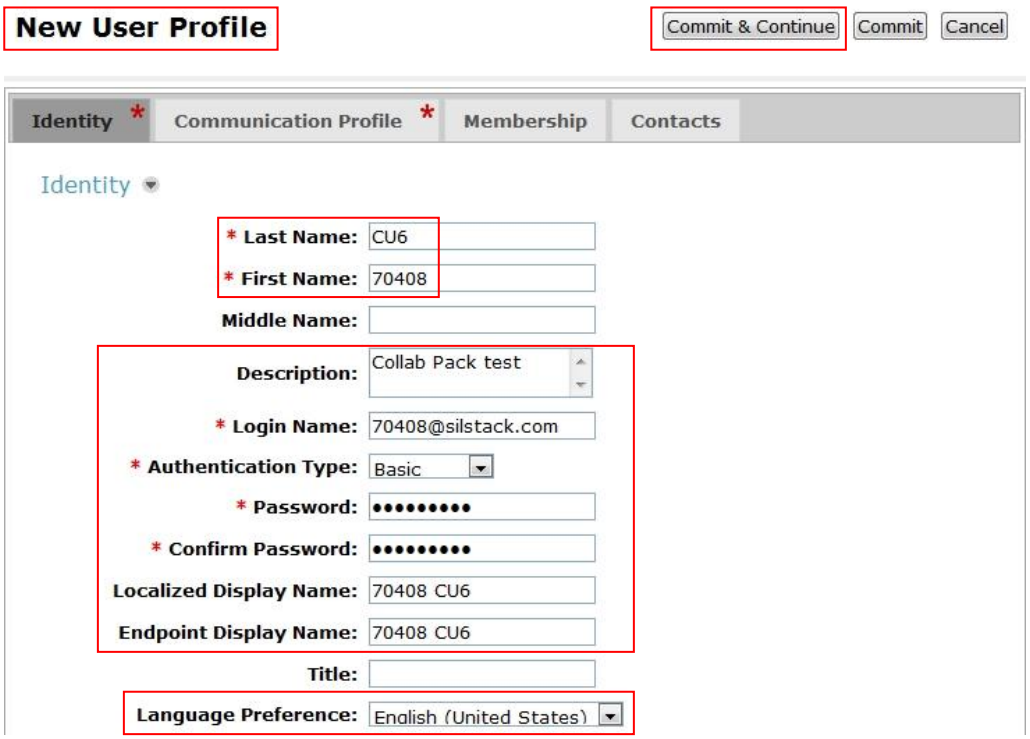


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In the **New User Profile** page which appears, under the **Identity** section, enter values for the following required attributes for a new user and use default values for remaining fields.

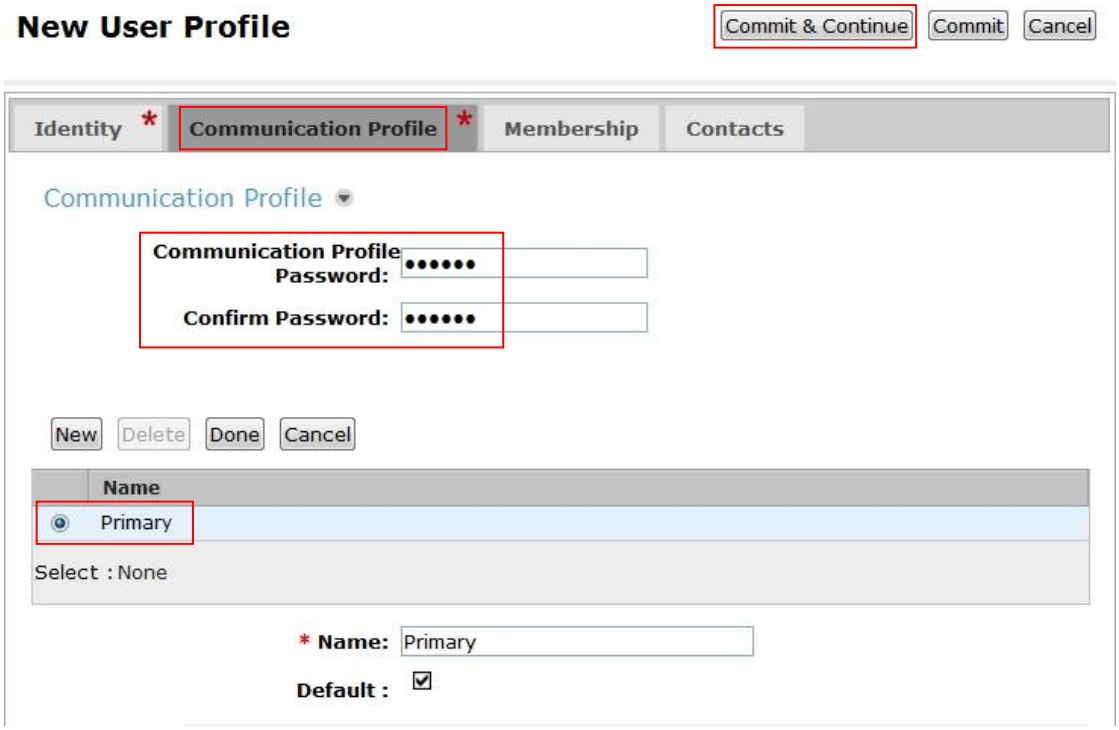
- **Last Name:** Enter last name of user (e.g. **CU6** as noted above in **Section 8.1**).
- **First Name:** Enter first name of user (e.g. **70408** as noted above in **Section 8.1**).
- **Description:** Optionally enter a description.
- **Login Name:** Enter using the format "**handle@<domain>**" where "**<domain>**" matches the domain from **Section 6.1** (e.g. **70408@silstack.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.



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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.

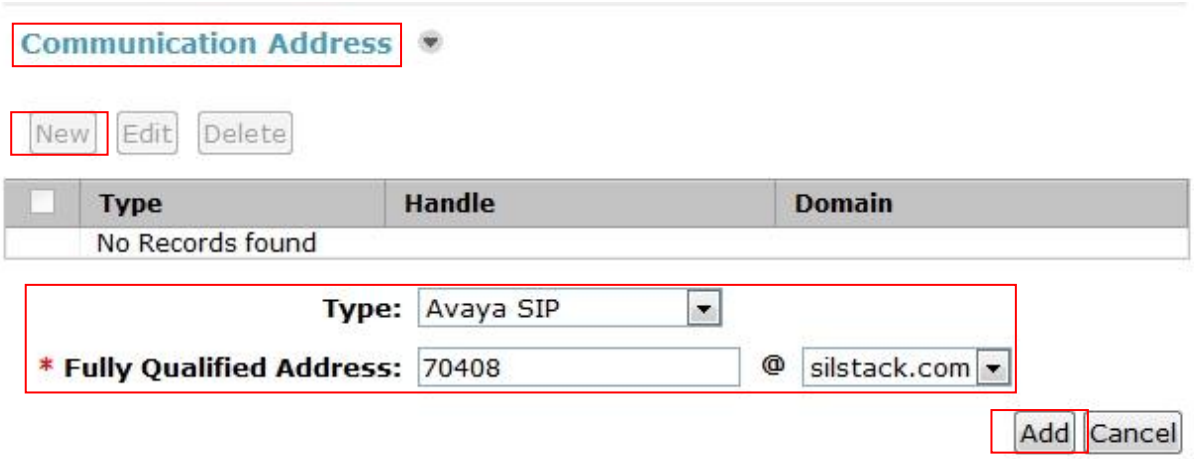


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Next**,** Expand the **Communication Address** sub-section and select **New** to define a **Communication Address** for the new user. Enter values for the following required attributes:

- **Type:** Select **Avaya SIP** from the drop-down menu.
- **Fully Qualified Address:** Enter same extension number as used for Login Name in Step 1 of this **Section 8.2**. **Note:** value is shown under the "Handle" field after the address is added.
- **Domain:** Verify value matches Domain name defined in **Section 6.1** (e.g. **silstack.com**).

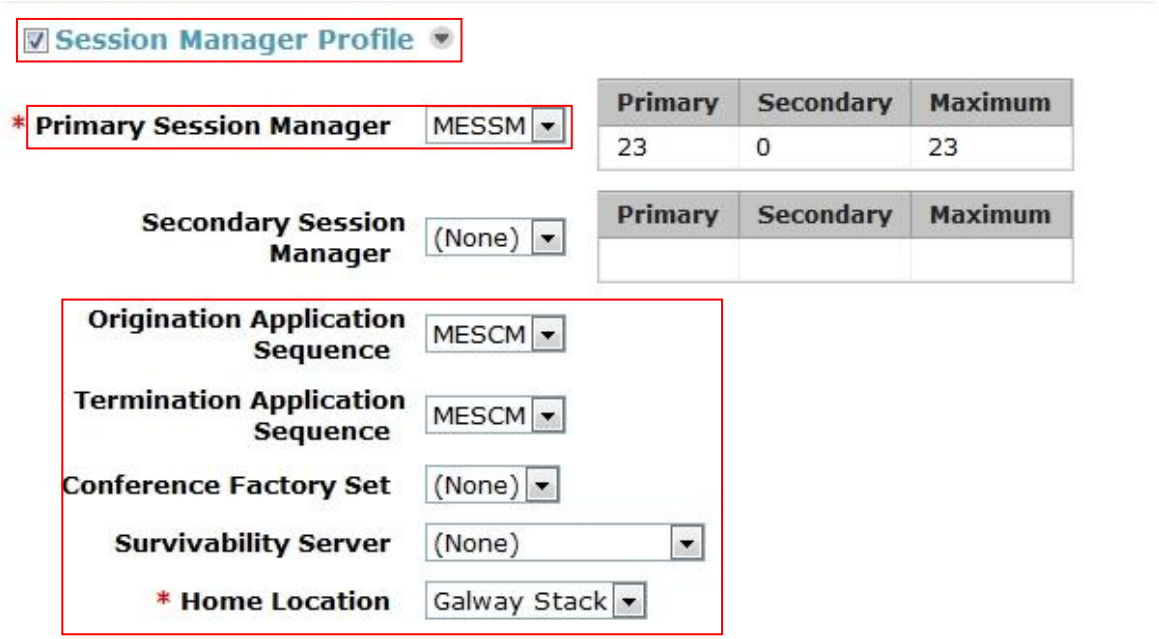
Click **Add** to save the Communication Address. **Note:** The Avaya XMPP communication address is added automatically to the communication profile because Presence Services is enabled as part of the Avaya Midsize Enterprise template. Telephony Presence will not display properly on a Collaboration client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



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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. **MESSM**).
- **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**).
- **Conference Factory Set** Retain the default value of **(None)**.
- **Survivability Server** Select **(None)** from drop-down menu.
- **Home Location** Select a Location from the drop-down list (e.g. **Galway Stack**).



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

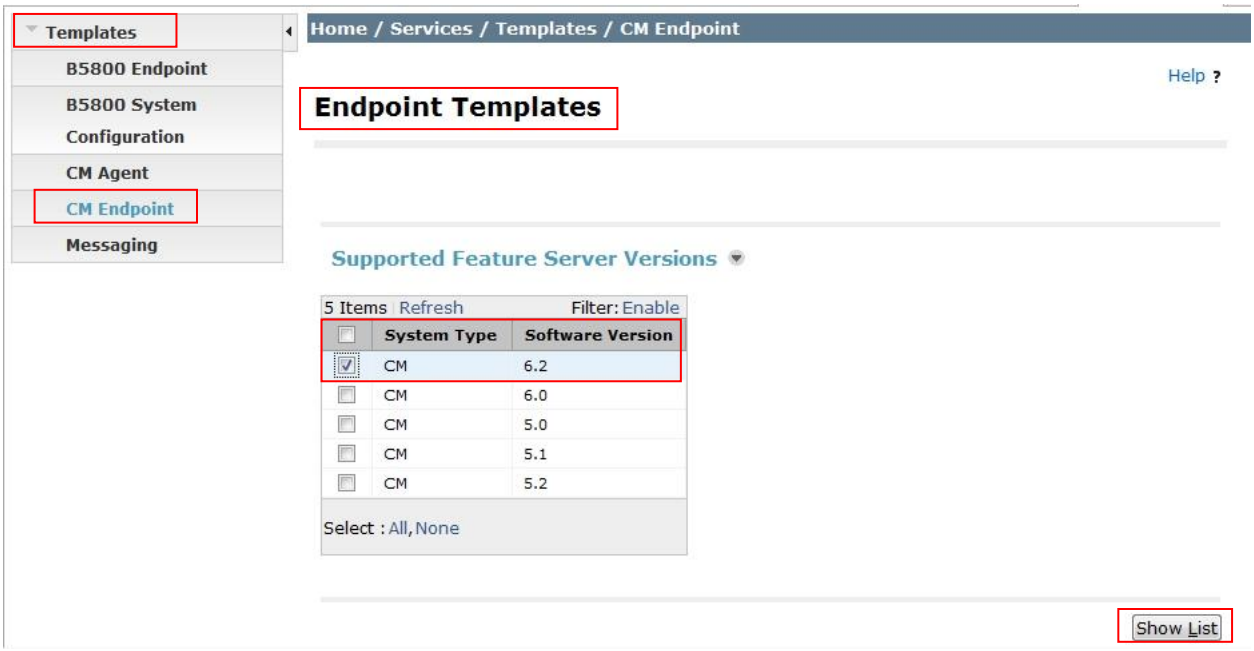
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Prior to creating the Communication Manager endpoint profile for a non-standard user / endpoint (e.g. One-X Mobile SIP for iOS or Flare Experience), it is recommended to create a customized template. There is no standard template for Flare Experience or one-X Mobile 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 6.2 page under the **Services** list, click on **Templates.**

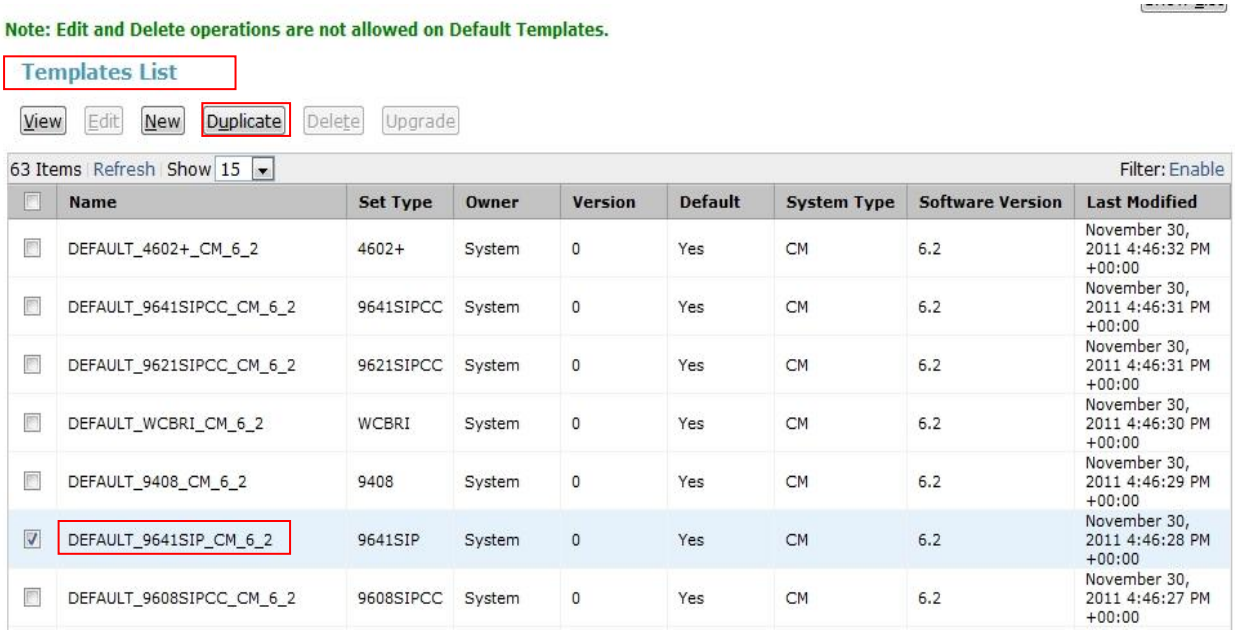


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Under the **Templates** list on the left navigation menu, click on **CM Endpoint**. In the **Endpoint Templates** check the box for **System Type** and **Software Version** as **CM 6.2**. Click on **Show List**.

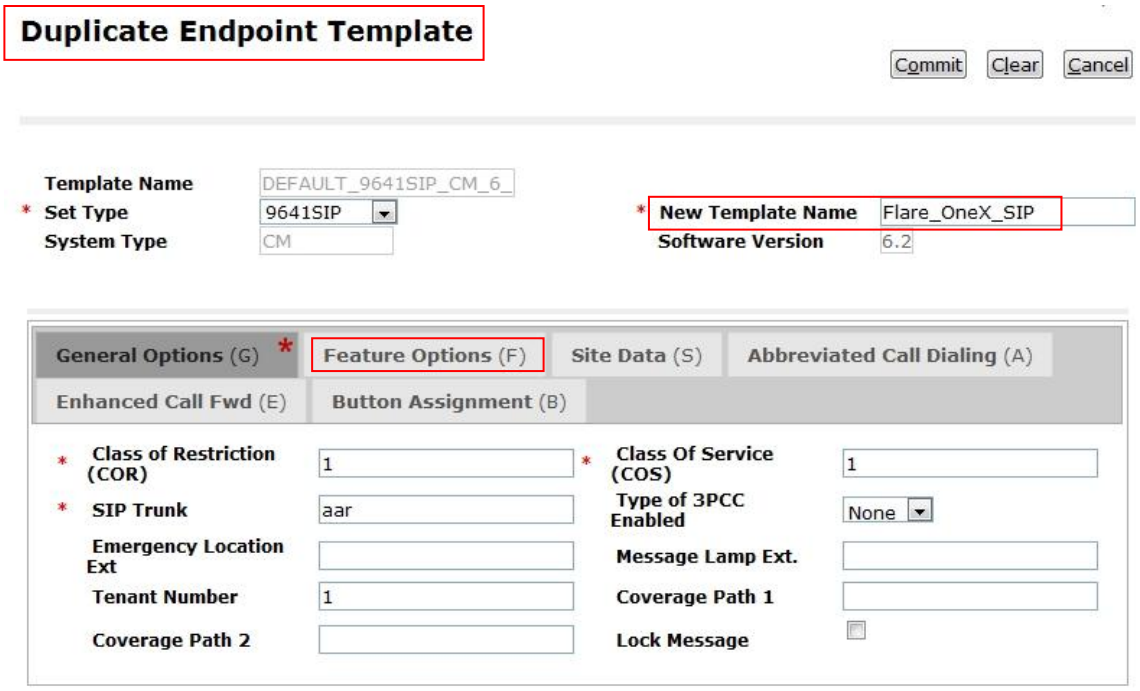


In the **Templates List** screen find the template called **DEFAULT_9641SIP_CM_6_2**, select the checkbox and the select the **Duplicate** button.

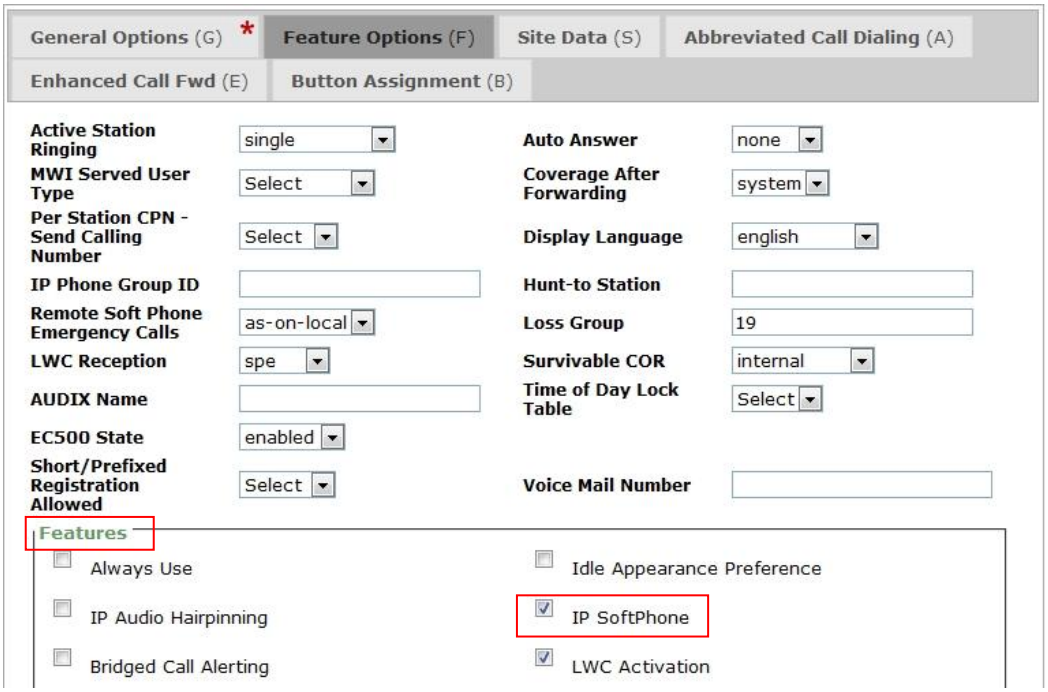


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In the next screen, **Duplicate Endpoint Template**, enter a name in the **New Template Name** box (e.g. **Flare_OneX_SIP**). Click on the **Feature Options (F)** tab.



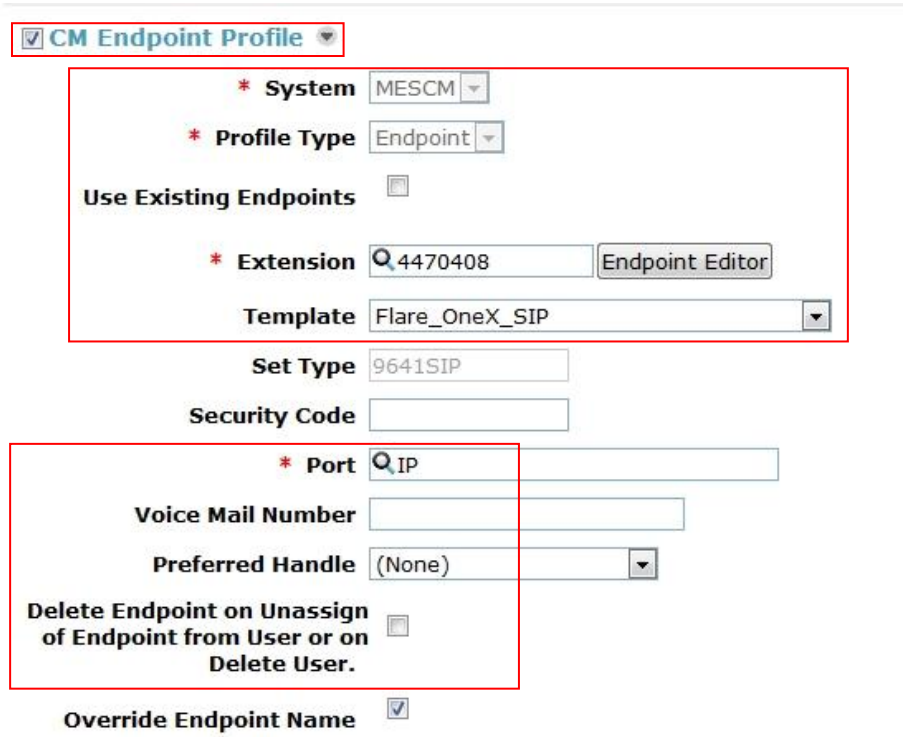
In the list of **Features**, tick the box for **IP Softphone**. Select Commit (not shown) to save the new template.



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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 new template just created called **Flare_oneX_SIP**
- **Security Code** Enter numeric value used to register the 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).
- **Delete Endpoint on Unassign of Endpoint from User or on Delete User** This is an optional tick box to automatically delete station when the Endpoint Profile is un-assigned from user.



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

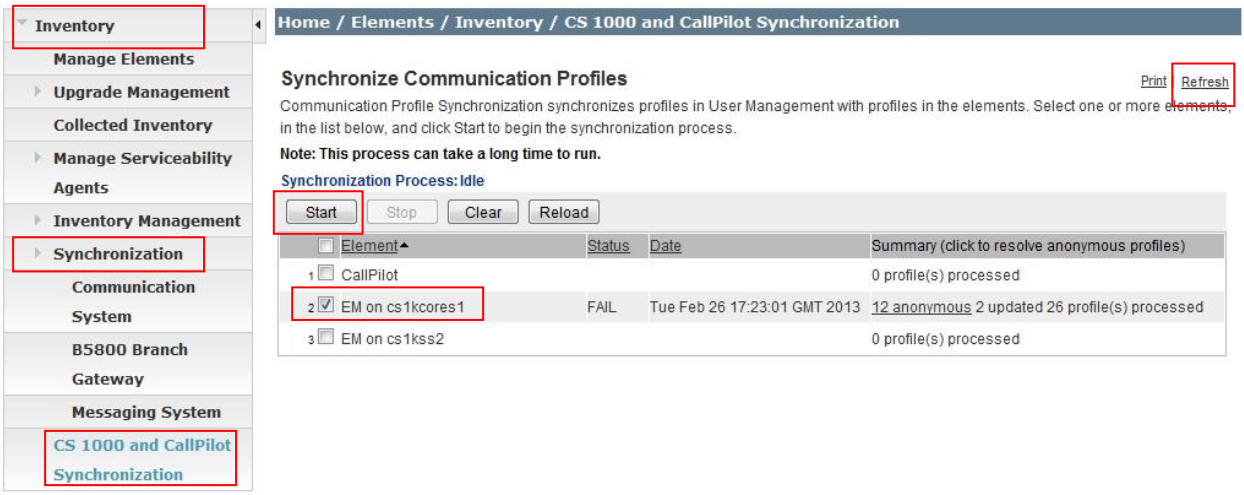
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8.3. Synchronize Communication Profiles

System Manager provides an account synchronization feature to synchronize profiles between the different elements of the solution e.g. CS1000, 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 CS1000 will import and synchronize all CS1000 users into their previously created System Manager Identity CS1000 Endpoint Profiles by matching each user's **First Name** and **Last Name**. After building the user identities, perform an on demand synchronization with CS1000. From the System Manager 6.2 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 CS1000 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 CS1000 user name match.



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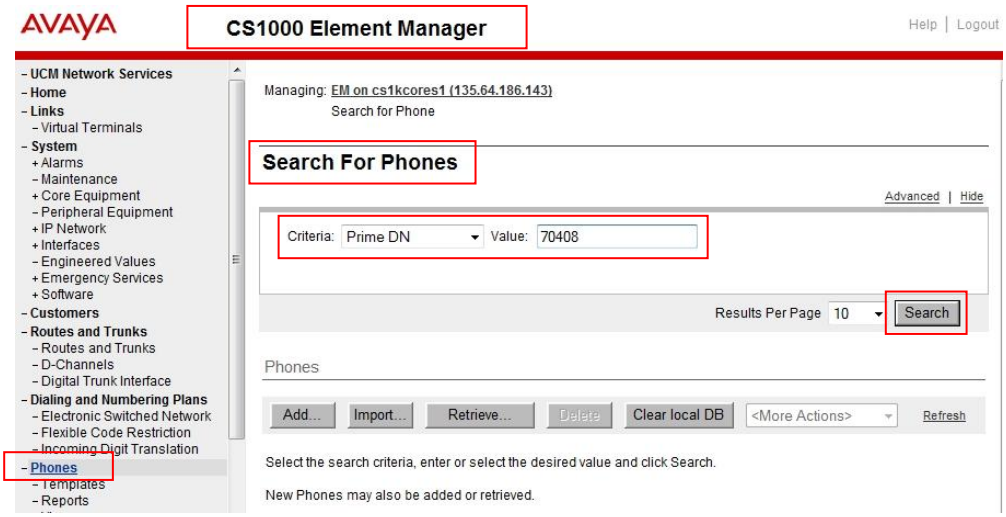
8.4. Personal Call Assistant Configuration (PCA)

The PCA feature is utilized to enable the twinning of a CS1000 with an Avaya SIP endpoint (e.g. Avaya Aura® Flare Experience) so that a "Converged User" may have simultaneous ringing, single number, single voicemail box and aggregated Presence. It is also used for Native users to give them a Prime DN on the CS1000 and a routing mechanism to direct calls across the SIP trunk to the Collaboration Pack. The steps below describe PCA configuration for an existing CS1000 user (e.g. Prime DN 70408) to enable that user to be converged with a collaborated endpoint. Follow same steps to configure a PCA profile for a new Native User (e.g. Prime DN 70080) - these steps are not shown here.

To achieve the simultaneous ringing a PCA must be configured per Converged User. When the CS1000 client endpoint receives an incoming call, the PCA sends call signaling to the Avaya client endpoint via a CS1000 SIP trunk to Session Manager to effectively "simultaneously ring" the Converged User's Avaya client endpoint. It is presumed that the PCA feature is licensed on the CS1000 and enabled in the CS1000 Customer Data Block. The following steps are required to configure a PCA for every CS1000 phone user:

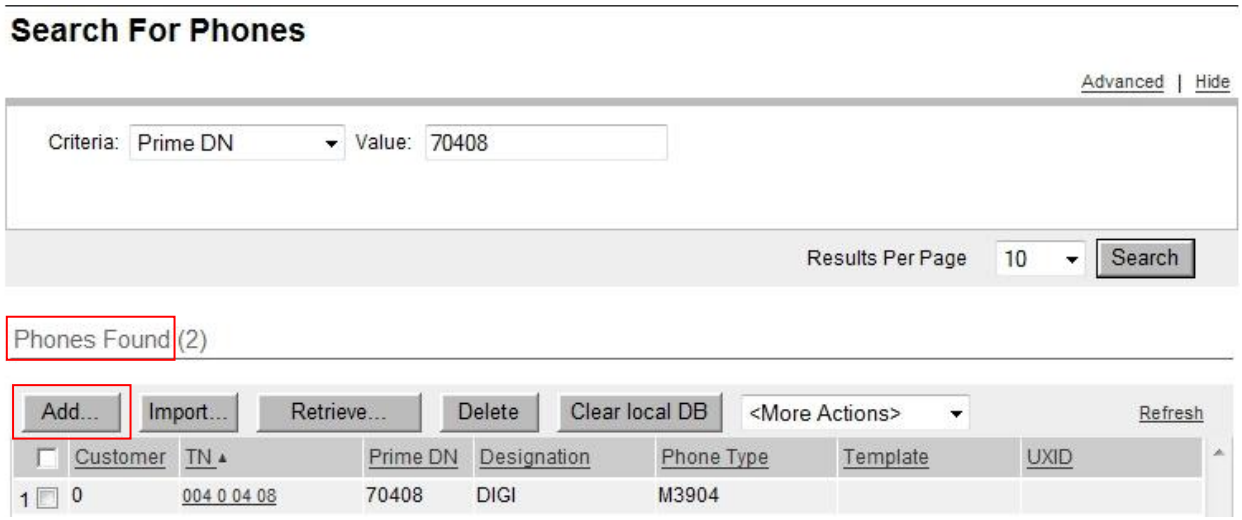
- The PCA shares the same primary DN on key 0 as the CS1000 endpoint client (e.g. 70408 for an existing CS1000 user or 70080 for a new native user).
- Configure key 0 as the Primary DN.
- Configure key 1 as HOT P key with the appropriate route prefix and DN as required to reach the twinned Avaya client endpoint.

To configure a PCA entry for an existing CS1000 user, first carry out a search for the Prime DN of that user using UCM Services and then add a PCA. Staying with the example used in this Application Note, CS1000 user Prime DN **70408** is shown. Using the UCM Services link in the System Manager main page, access the **CS1000 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 CS1000 user (e.g. **70408**). Click on **Search** button.

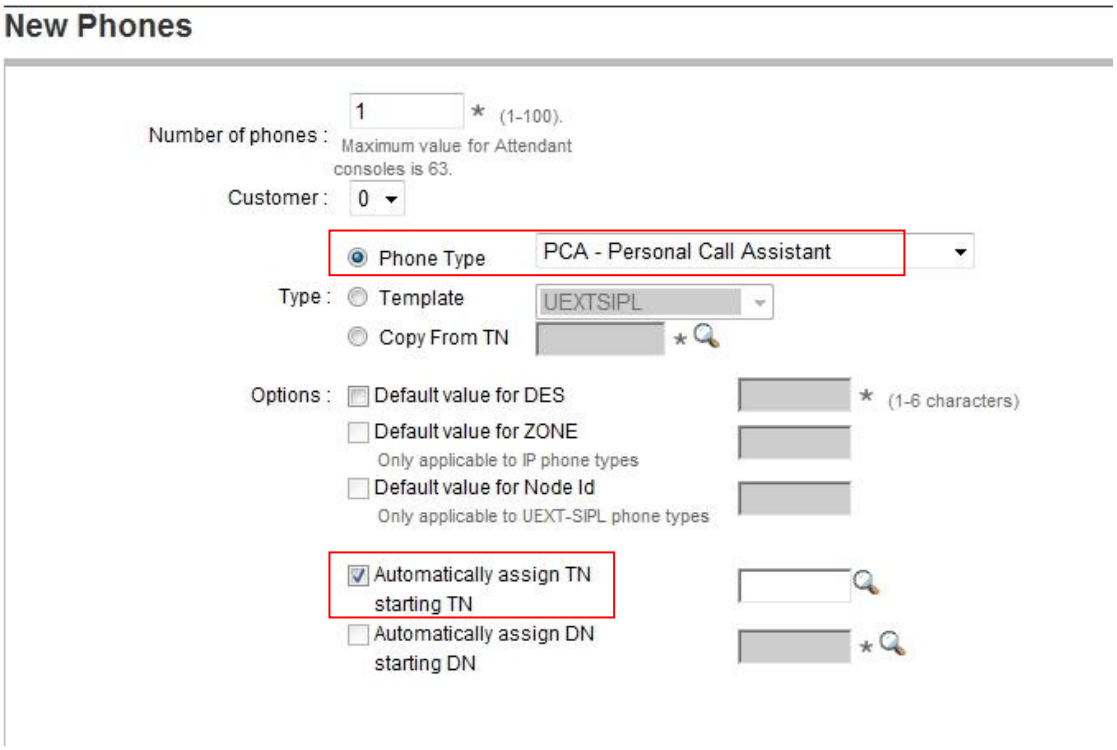


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The result of the search comes back and displays **Phones Found**. Next, select the **Add** button.

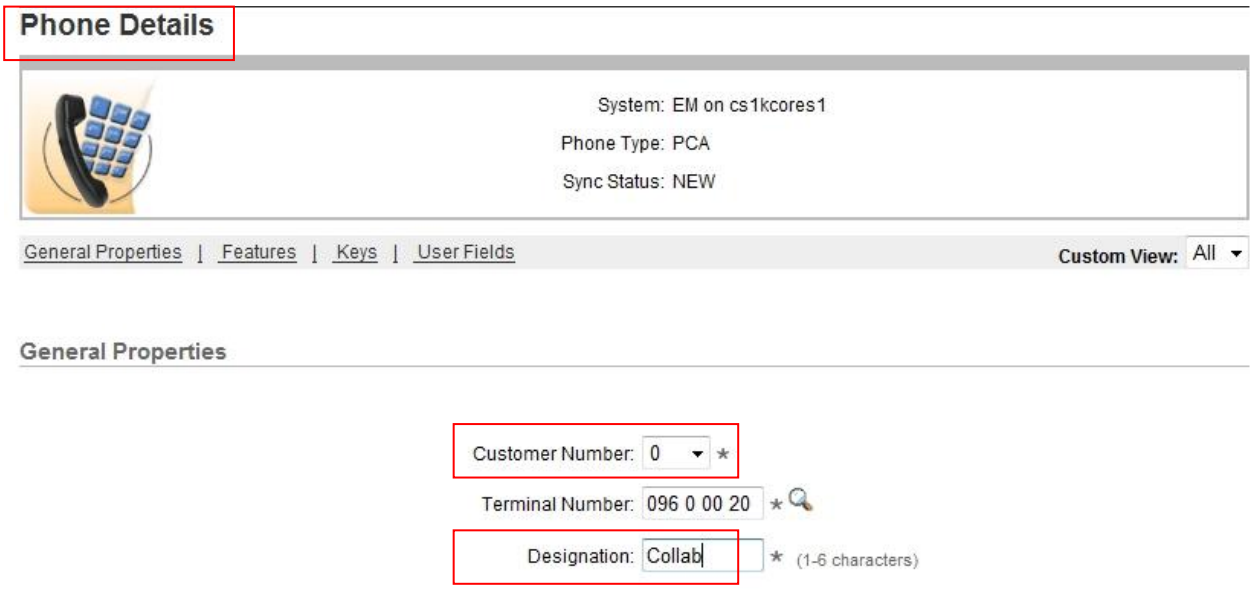


For **Phone Type** select **PCA-Personal Call Assistant** from the drop down menu. Next check the box to **Automatically assign TN starting TN**. Scroll down and select the Preview button (not shown).

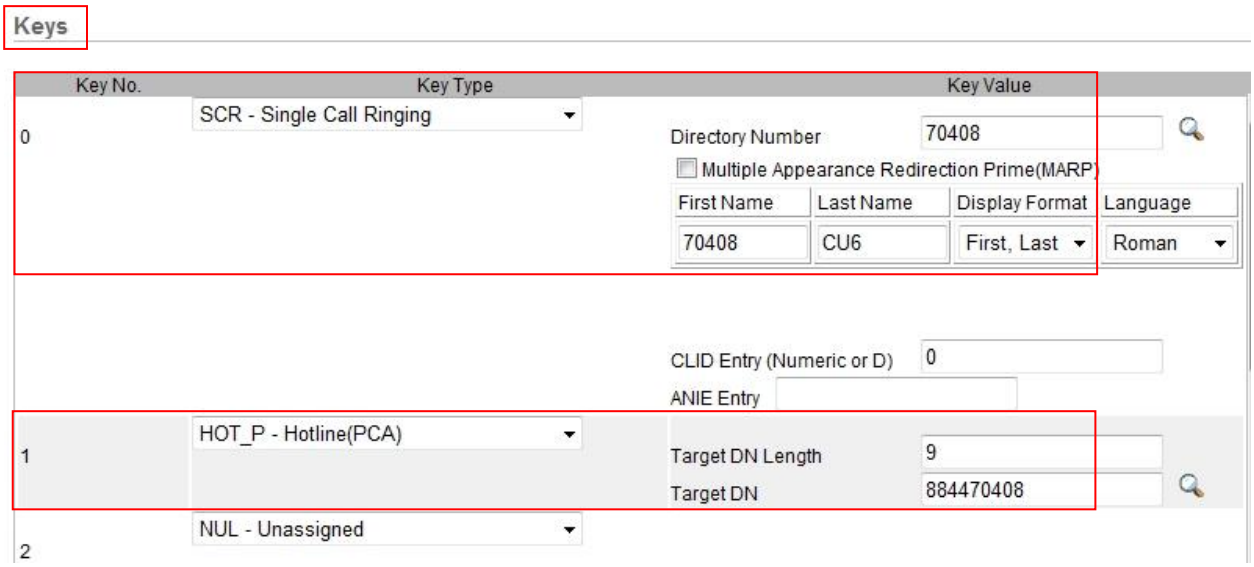


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The Phone Details screen appears. Enter a CS1000 **Customer Number** (e.g. **0**) and **Designation** (e.g. **Collab**).



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. **9**) and the **Target DN** of the Collaborated endpoint (e.g. **884470408**).



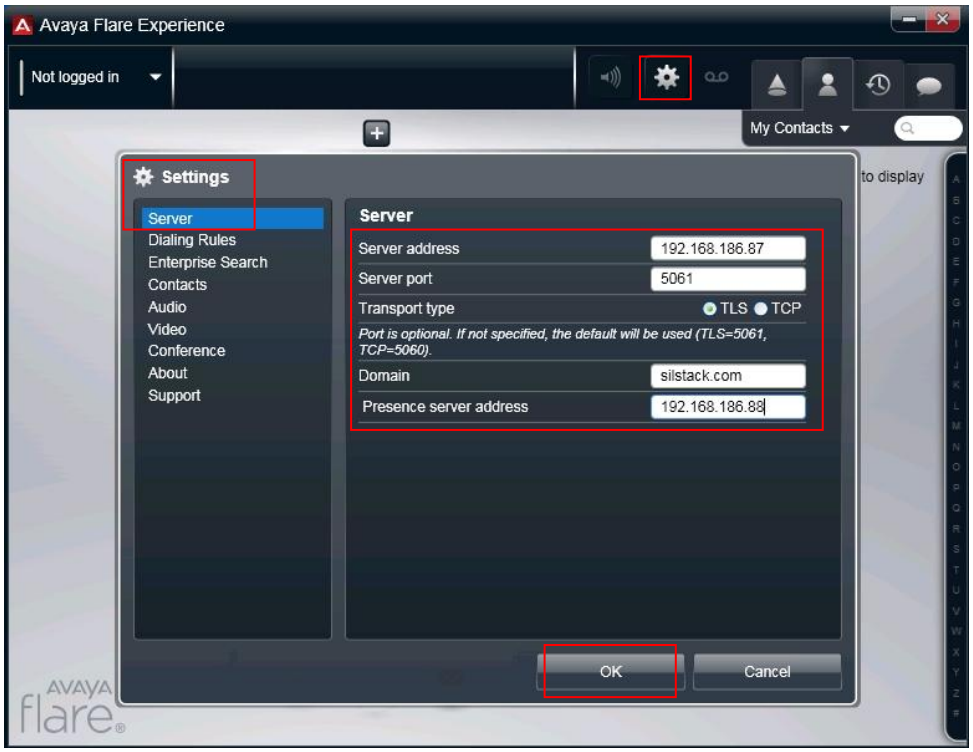
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Select Commit (not shown) to save changes. When this is done, preform the account re-synchronization of the CS1000 users to System Manager again using instructions in **Section 8.3.1**.

8.5. Manual Configuration of Avaya SIP Clients

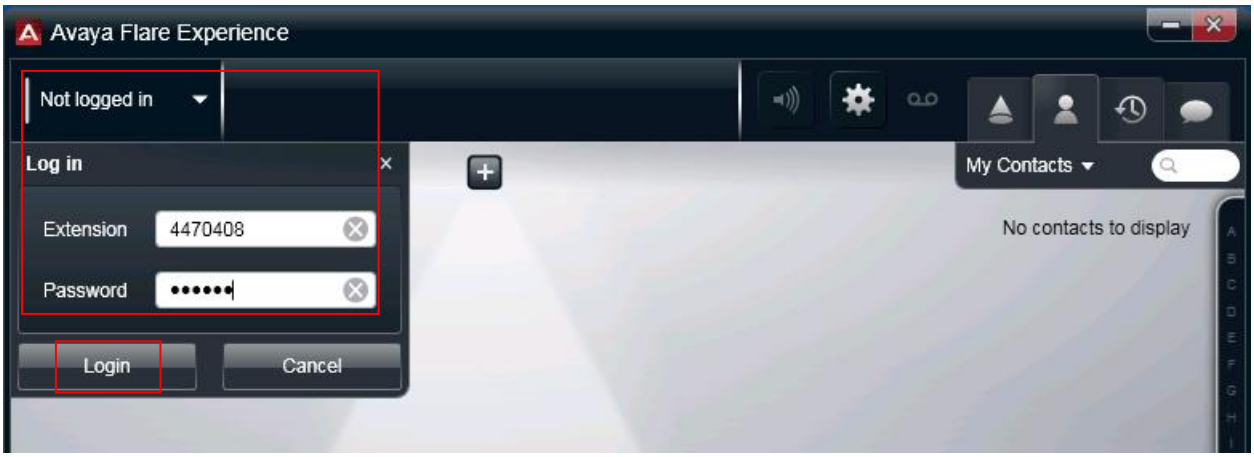
This section shows the steps required to manually configure an Avaya Flare Experience for Windows as an example of configuring a Collaboration endpoint / user to register to Session Manager.

It is presumed that the Avaya Flare Experience for Windows has been previously installed and the application has been launched. Click on the star icon to open the **Settings** window. From the left hand list, select **Server**. In the main Server window, enter the **Server address** as the Session Manager asset (e.g. **192.168.186.87**). For **Transport type**, click on the TLS button and the Server Port should automatically change to 5061. In the **Domain field**, enter the SIP domain used for the solution (e.g. **silstack.com)**. For the **Presence server address** enter the IP address of the Presence Server (e.g. **192.168.186.88)**. Click on OK to save the settings.

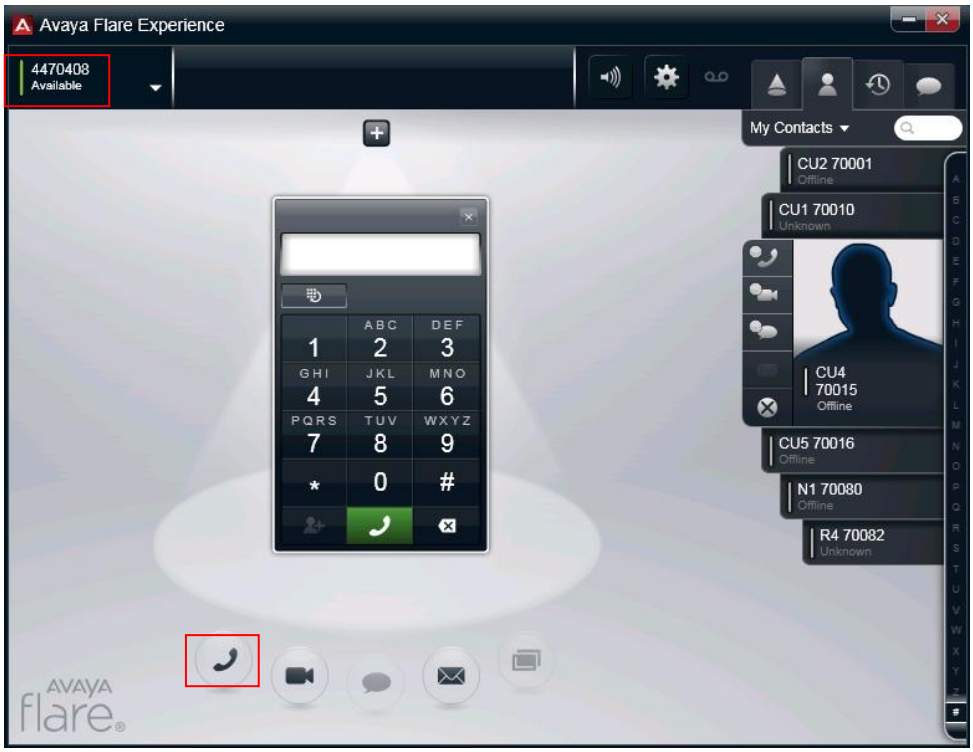


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To login, the user must provide the required login credentials of **Extension** and **Password**. In this example, to login user "70408, CU6", click on the down-arrow beside the **Not logged in** text on the client. The **Log in** drop-down window appears. Because the client is logging into the ME system, the full extension number including the prefix must be used. So enter **4470408** as the **Extension** and for **Password** enter **123456**. Click on **Login** to complete the login process.



The Avaya Flare Experience for Windows client is now logged in with presence status showing as **Available** and becomes the collaborated / converged endpoint for the CS1000 user with DN 70408. Clicking on the **call icon** on the main screen brings up the keypad to allow the user to initiate calls.



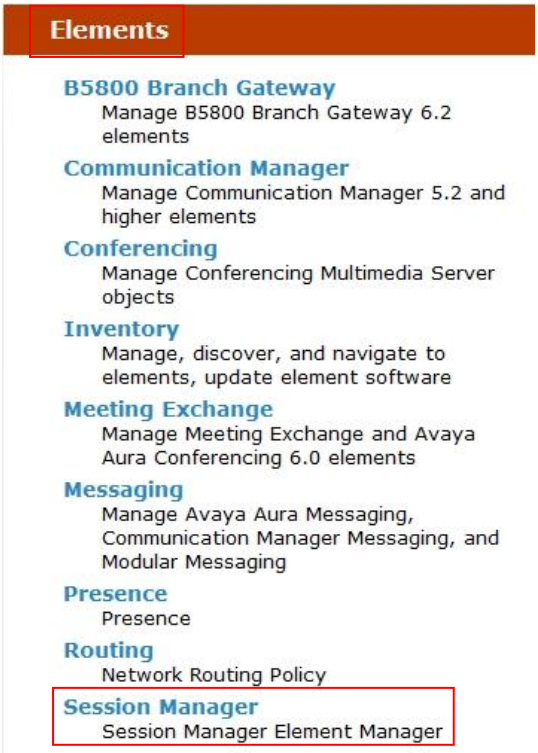
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9. Verification Steps

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

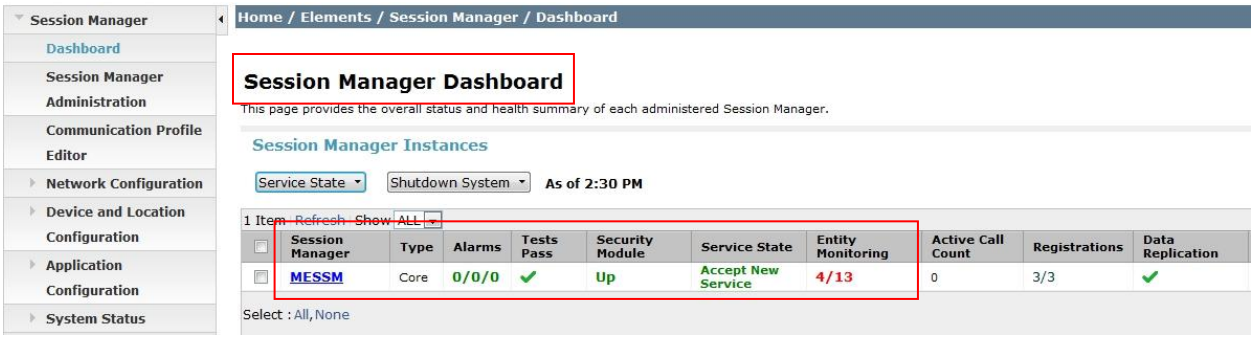
To verify Session Manager operational status, navigate from the main System Manager 6.2 page under the **Elements** list, click on **Session Manager**.



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

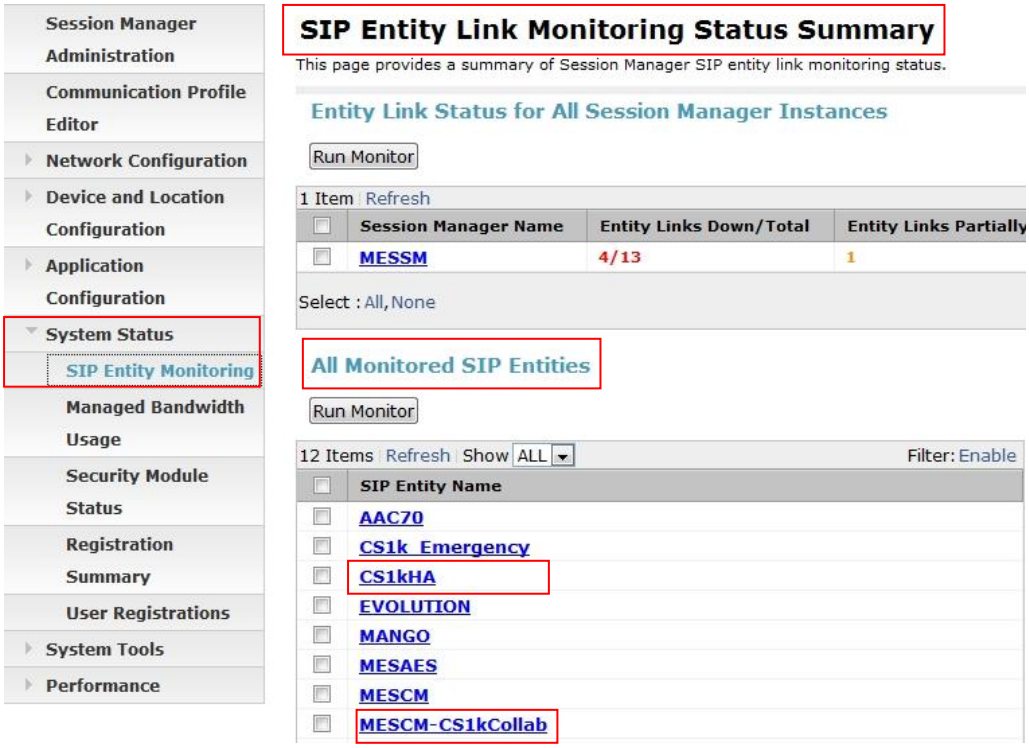
- **Alarms** should show as 0/0/0 to indicate no alarms present,
- **Test Pass** should have a green tick-mark,
- **Security Module** should show as **Up**,
- **Service State** should show as **Accept New Service**,
- **Entity Monitoring** should ideally show a count indication of 0 entity down links / 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).



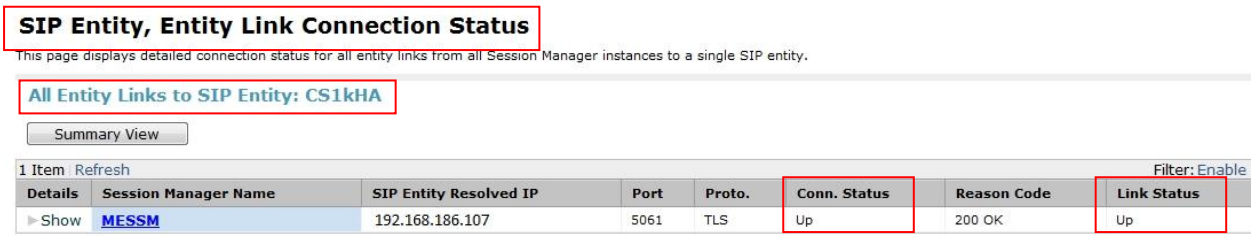
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9.2. Verify Avaya Aura® Session Manager Entity Links Status

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 CS1000 Collaboration Pack 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**.



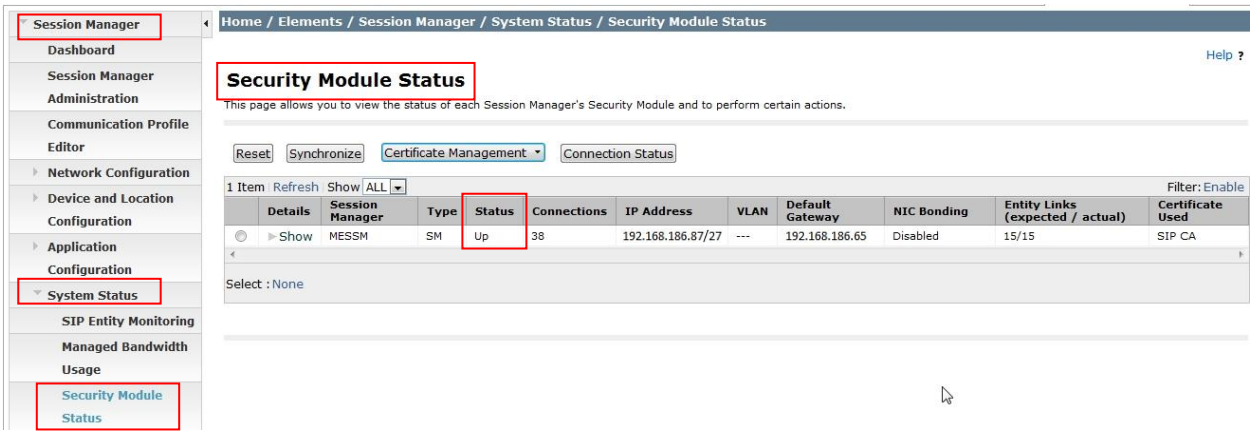
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Similarly, from the SIP Entity Link Monitoring Status Summary page, click on the MESCM-CS1kCollab link next to check its status (not shown). The **SIP Entity, Entity Link Connection Status** window appears. It shows **All Entity Links to SIP Entity: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

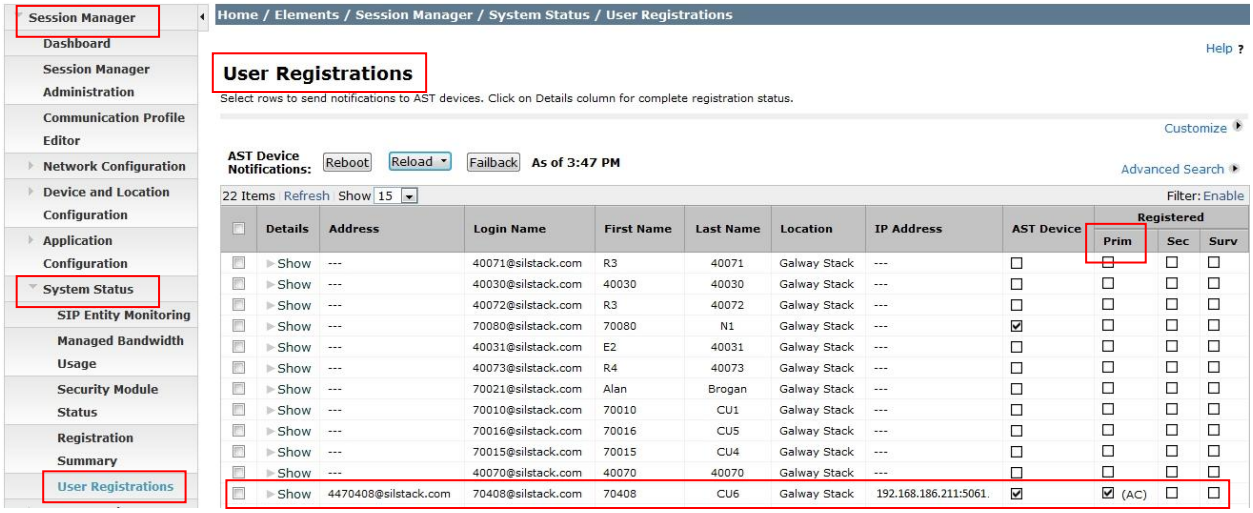
Next verify the Session Manager Security Module status. From the **Session Manager** drop-down list on the left hand side, click on **System Status** and then on **Security Module Status**. In the **Security Module Status** window, verify the **Status** column displays **Up** as shown below.



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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.



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10. Conclusion

This Application Note describes the configuration and test results for Collaboration Pack 1.1 for CS1000 Release 7.6.

The Collaboration Pack 1.1 solution is made up of the Avaya Aura® Midsize Enterprise system Release 6.2 and Avaya Communication Server 1000E Release 7.6 with CallPilot Release 5.1. A stepping stone for Avaya Communication Server 1000E installed base migration to a full Avaya Aura solution, the Collaboration Pack 1.0 on Avaya Communication Server 1000E Release 7.5 introduced the concept of associating a collaboration client on Avaya Aura® Midsize Enterprise Server with an existing Avaya Communication Server 1000E desk phone via PCA ("twinning"). Collaboration Pack 1.1 now extends the number of twinned collaboration clients and aligns with Avaya Communication Server 1000E Release 7.6. Release 1.1 defines 4 types of users:

- **Non-Converged Users**: Existing CS1000 users who continue to use their CS1000 desk phone with no association to any client on the ME Collaboration Pack.
- **Converged Users**: Existing CS1000 users who continue to use their CS1000 desk phone and are also associated, via PCA on the CS1000 side, with a client on the ME Collaboration Pack. Some of these ME clients may be remote users who are connected into the Enterprise network over the WAN via ASBCE.
- **Native User**: A user on the ME Collaboration Pack that has no corresponding CS1000 desk phone. Some of these ME clients may be remote users who are connected into the Enterprise network over the WAN via ASBCE.
- **Remote User**: Remote Users registered to Avaya ME via the optional ASBCE. These can be considered a subset of the Converged and Native user groups.

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

Non Converged Users:

- 11xx IP (UNISTIM) deskphone
- 12xx IP (UNISTIM) deskphone
- 39xx Digital deskphone

Converged Users:

- 11xx IP (UNIStim) deskphone – CS1000
- 12xx IP (UNIStim) deskphone – CS1000
- 39xx Digital deskphone – CS1000
- Flare® Communicator for iPad – ME
- One-X Mobile (SIP) for iOS – ME
- Flare® Experience for iPad – ME
- Flare® Experience for Windows XP – ME
- Flare® Experience for Windows 7 – ME
- Remote Flare® Experience for iPad registered to ME via ASBCE
- Remote Flare® Experience for Windows registered to ME via ASBCE
- Remote one-X® Mobile SIP for iOS registered to ME via ASBCE

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Native Users:

- B179 Conference SIP phone
- One-X Communicator (SIP) with Audio provided by a 96x1 SIP phone
- One-X Communicator (SIP) with Audio provided by a 96x1 SIP phone and extended with EC500 to an Avaya one-X® Mobile Lite.
- Avaya one-X® Mobile with Client Enablement Services (CES)
- Avaya one-X® Deskphone SIP on 96X1
- Remote Avaya one-X® Deskphone SIP on 96X1 registered to ME via ASBCE.

All testing was successful with the exception of those issues and limitations documented in **Section 2.2**.

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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 CS1000 Collaboration Pack solution.

A2. Configuring Secure SIP Connectivity using Transport Layer Security (TLS) between Avaya Aura® Communication Manager R6.2, Avaya Aura® Session Manager R6.2 and Avaya Communication Server 1000E R7.6.

A3. Application Notes for Configuring Avaya one-X® Mobile SIP Communicator for iOS with Avaya Aura® Session Manager R6.2, Avaya Aura® Communication Manager R6.2 as an Evolution Server and Avaya Aura® Messaging R6.1

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

Specific CS1000 Release 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® documentation are also listed below for reference:

- ME Intelligent Workbook,
- Overview of Avaya Aura® Solution for Midsize Enterprise, Release 6.2,
- Implementing Avaya Aura® Solution for Midsize Enterprise Template Release 6.2,
- 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,
- Avaya Flare Overview and Planning,
- Administering Avaya Flare® Experience for Windows,
- Administering Avaya Flare® Experience for iPad,
- Avaya one-X Mobile for IP Office Administration Guide.

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Avaya Aura® Presence Services documentation relevant to this sample configuration are listed below for reference:

- Presence Services with Communication Server 1000 (NN43001-141)
- Implementing Avaya Aura® Presence Services Release 6.1
- Administering Avaya Aura® Presence Services Release 6.1

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 CS1000 Collaboration Pack solution components of Avaya Aura® Conference 7.0 and ASBCE are given below. These documents can be obtained from http://support.avaya.com.

Avaya Aura® Conference 7.0 documentation references:

- Avaya Aura® Conferencing 7.0 Overview and Specification
- Avaya Aura® Conferencing 7.0 Planning and Design
- Avaya Aura® Conferencing 7.0 Security
- Avaya Aura® Conferencing 7.0 Accounting Records Reference
- Avaya Aura® Conferencing 7.0 Alarms and Logs Reference
- Avaya Aura® Conferencing 7.0 Operational Measurements Reference
- Avaya Aura® Conferencing Collaboration Agent Quick Reference
- Deploying Avaya Aura® Conferencing 7.0
- Administering Avaya Aura® Conferencing 7.0
- Maintaining and Troubleshooting Avaya Aura® Conferencing 7.0
- Using Avaya Aura® Conferencing Collaboration Agent.

Avaya Session Border Controller for Enterprise (ASBCE) documentation references:

- Installing Avaya Session Border Controller for Enterprise
- Upgrading Avaya Session Border Controller for Enterprise
- Administering Avaya Session Border Controller for Enterprise.

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