Avaya Solution and Interoperability Test Lab

Application Notes for Polycom RMX 4000 Version 7.2.2.16 and Avaya Communication Server 1000 Release 7.5 and Avaya Aura® Session Manager 6.1 – Issue 1.1

Abstract

These Application Notes describe a solution comprised of SIP interoperability for audio and video conferencing between Polycom RMX 4000 Multipoint Control Unit Version 7.2.2.16 and Avaya Communication Server 1000 Release 7.5 and Avaya Aura® Session Manager Release 6.1.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

The objective of this compliance test was to validate interoperability of the Polycom RMX 4000 Multipoint Control Unit (MCU) voice and video conferencing system with the Avaya Communication Server 1000 PBX System Release 7.5 that includes Call Sever, Signaling Server SIP Gateway, and SIP Line server. Endpoints used were IP Unistim Phones, Digital and SIP Phones. The testing used the Avaya Aura® Session Manager for routing calls between both Communication Server 1000 and Polycom RMX 4000.

2. General Test Approach and Test Results

The general test approach verified the RMX 4000 conferencing system is able to work with Avaya Communication Server 1000 and Avaya Session Manager by providing video and audio conference. All test cases were manually executed and verified by both Avaya and Polycom engineers to make sure it is working as expected.

2.1. Interoperability Compliance Testing

The focus of this testing was to verify the SIP interoperability between the Communication Server 1000 and RMX 4000 for audio and video conferencing conducted by the Polycom RMX 4000 MCU conferencing system, calls were routed between both the systems by the Session Manager.

There are two scenarios for calls coming in to the RMX 4000, the first scenario is to use the Through-dial feature of CallPilot to dial in to the RMX 4000, and the second is to call direct to the RMX 4000 from the Avaya endpoints connected to the CS1000.

The following test areas were practiced in the compliance testing:

- Verification of RMX 4000 features: creating new conferences, inviting participants into conference by dialing-out, Call Park, Hold/Retrieve, Supervised/Un-supervised transfer, Ad-hoc conference creations, and Ad-hoc conference scenarios.
- Verification of RMX 4000's features working as usual as using the through-dial feature of CallPillot to join conference call in the RMX 4000.

2.2. Test Results

The compliance testing was successful and all objectives were verified and met. All test cases were executed and passed.

Observation: The CallPilot Messaging system doesn't support video calling, only audio calls therefore video calls made from a video endpoint to the CallPilot Messaging system which then continues dialling to the RMX 4000 is only able to join a conference with audio, even if the conference supports video. The video endpoint can join in a video conference in the RMX 4000 if they directly call to the RMX 4000 and not go through the CallPilot.

2.3. Support

For technical support for the Polycom RMX 4000 MCU system, and Polycom products in general, please refer to www.polycom.com . On the Polycom website support hotline numbers for specific country will be found.

3. Reference Configuration

Figure 1 illustrates the network configuration used during the compliance testing between the Avaya Communication Server 1000, Avaya Aura Session Manager and the Polycom RMX 4000.

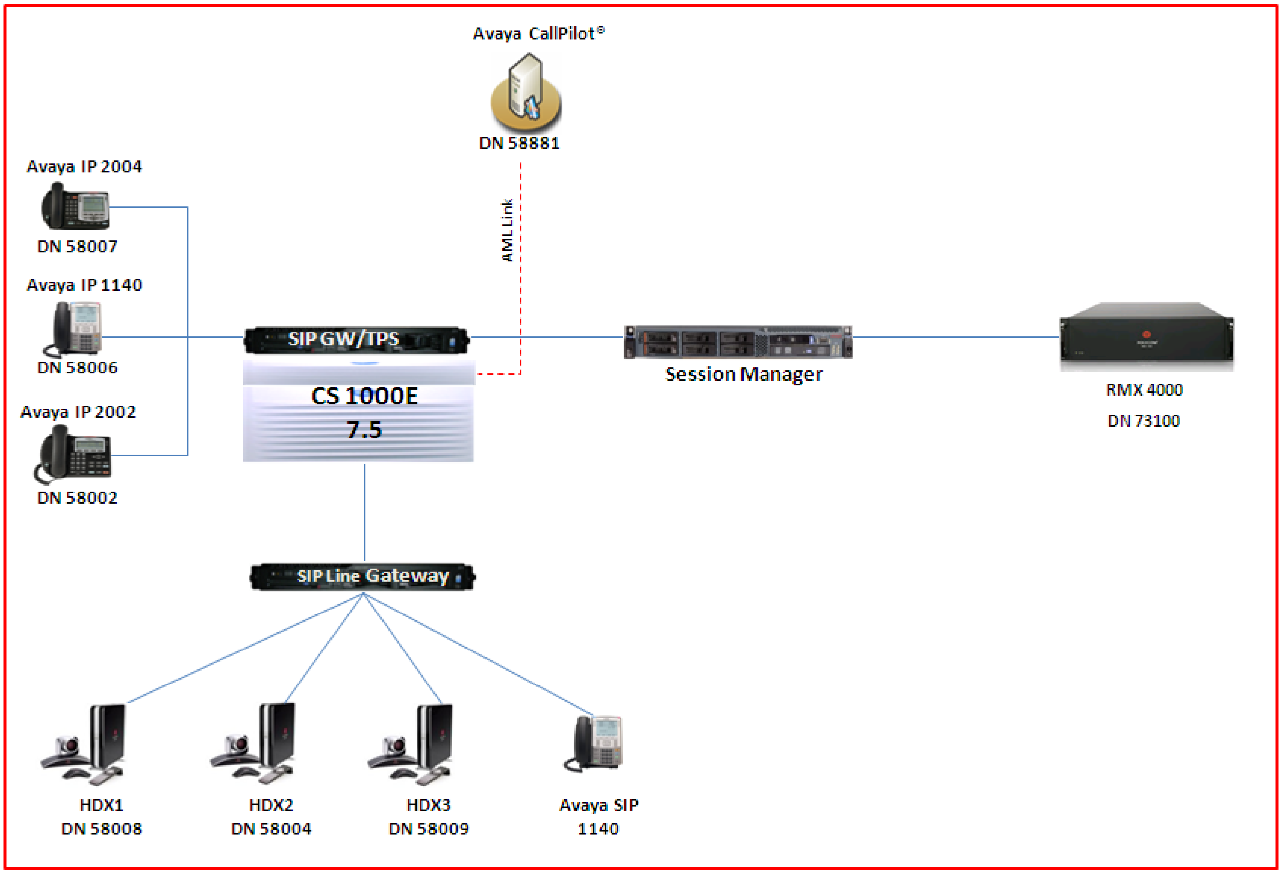

Figure 1: Network Diagram of the Tested Configuration

4. Equipment and Software Validated

The following equipment and software were used during the lab testing:

System	Software Version
Avaya Communication Server 1000E	Call Server (CPPM): 7.50 Q Signaling Server (CPPM): 7.50 Q
Avaya Aura® Session Manager	6.1.1
Avaya IP Phone 1140E Avaya IP 2004P2 Avaya IP 2002P2 Avaya Digital M3905	0625C7F 0692D93 0604DCN Flash: 9.0 P0 L1.8
Avaya SIP 1140	04.01.13.00
Avaya CallPilot® Messaging	05.00.41
Avaya CallPilot Application Builder	05.00.41
Polycom RMX 4000	7.2.2.16
Polycom HDX 8000	2.20 SP5

5. Configure Avaya Aura® Session Manager

This document assumes that the Session Manager was properly installed and configured as per the product documents; this section provides the steps on how to provision the Session Manager working with the CS1000 and Polycom RMX 4000. For more information about how to install and configure Session Manager, please refer to **Section 11 [1]**.

The following summarizes the tasks that need to be done in the Session Manager:
- Configure SIP Domain.
- Configure Locations.
- Configure SIP Entities (for CS1000 and Polycom RMX 4000).
- Configure Entities
- Configure Routing Policy.
- Configure Dial Pattern.

5.1. Configure SIP Domain

To configure a SIP Domain in the Session Manager, log in to the System Manager as shown in **Figure 2**. Enter the username "admin" and its password and then click on the **Log On** button to log in the System Manager.

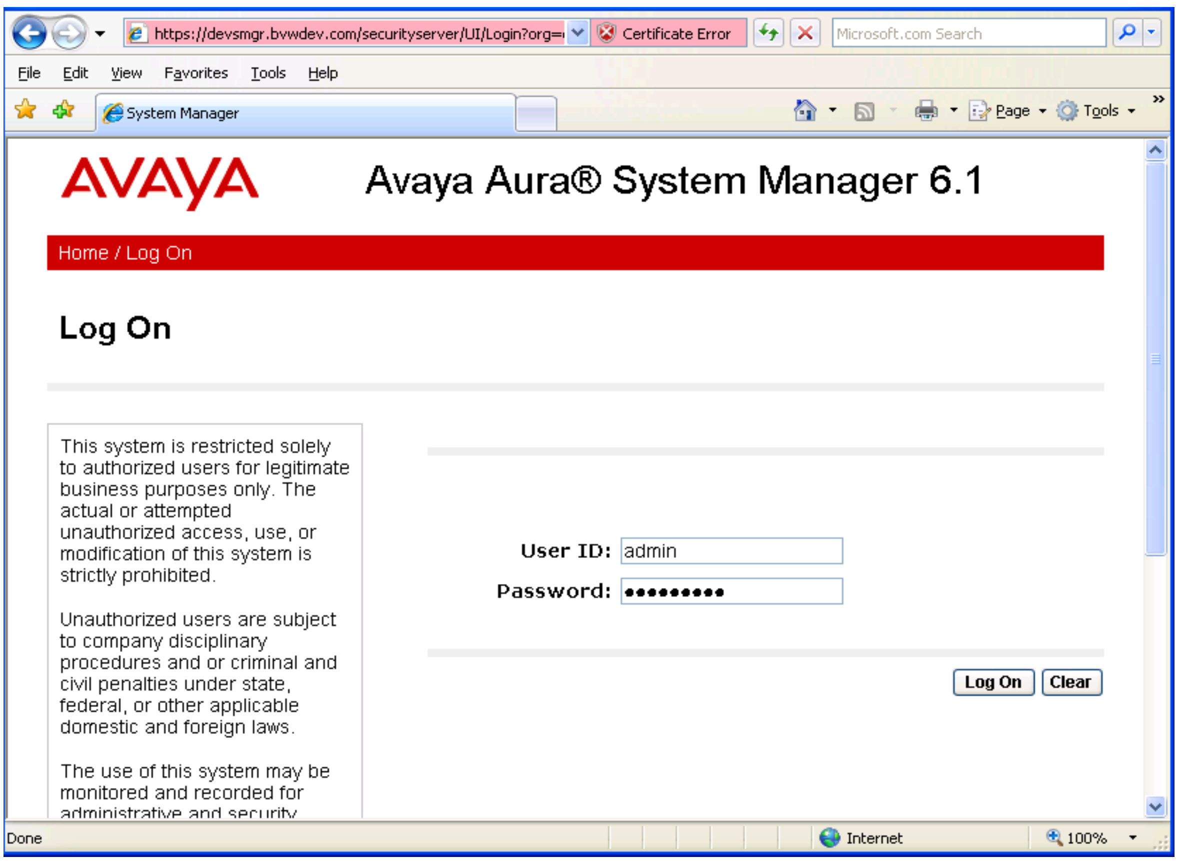

Figure 2: Log On window of System Manger

The homepage of System Manager appears as shown in **Figure 3**.

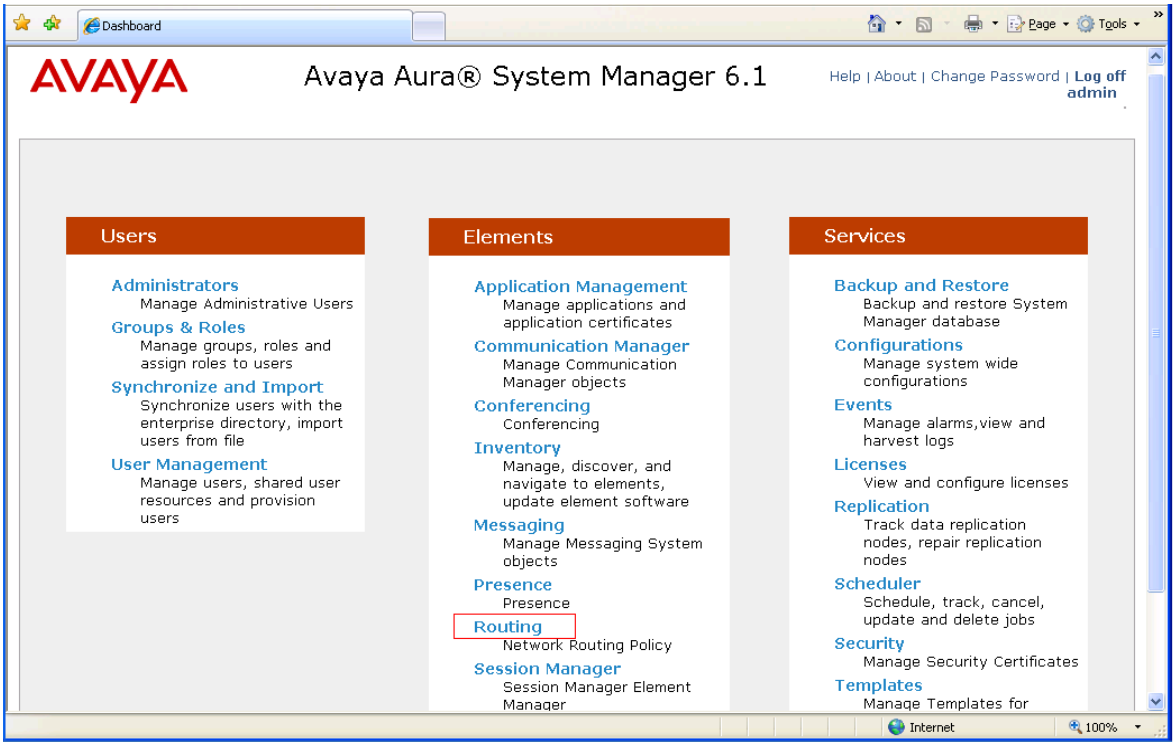

Figure 3: Home page of System Manager

Click on the **Routing** link under the **Elements** column, the **Routing** page appears as shown in **Figure 4**

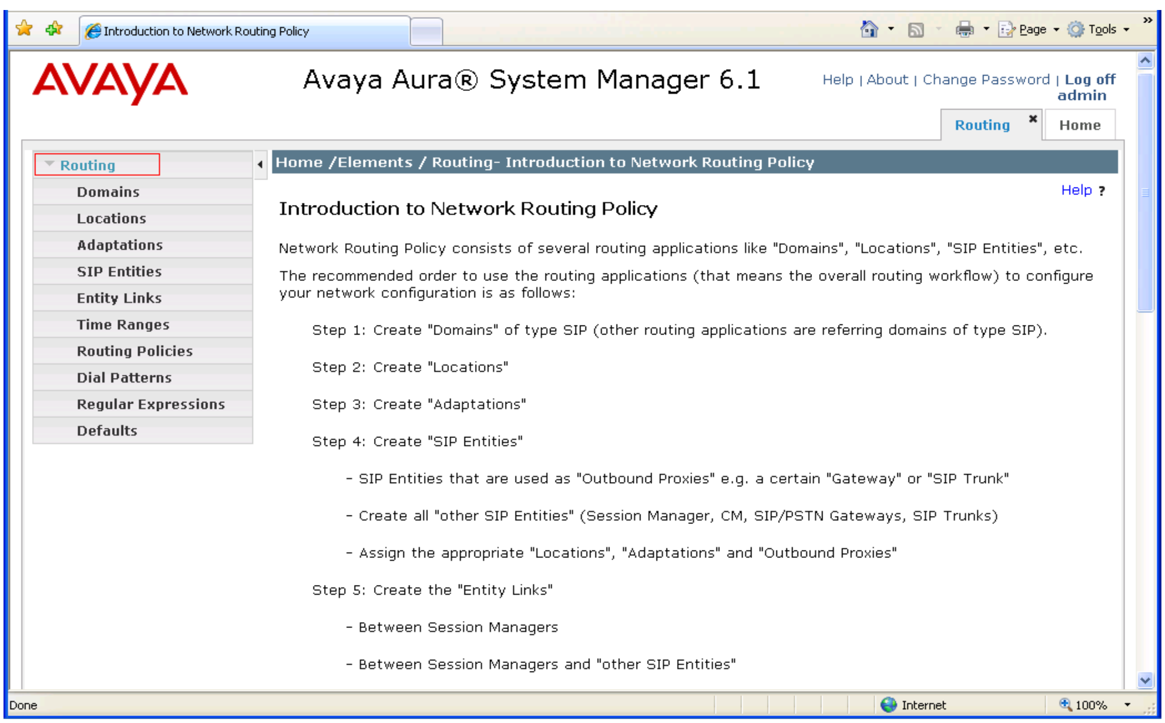

Figure 4: Routing – Introduction to Network Routing Policy

On the left-hand side of the **Routing** page, click on the **Domains** tab, the **Domain Management** appears in the right-hand side of the **Routing** page as shown in **Figure 5**.

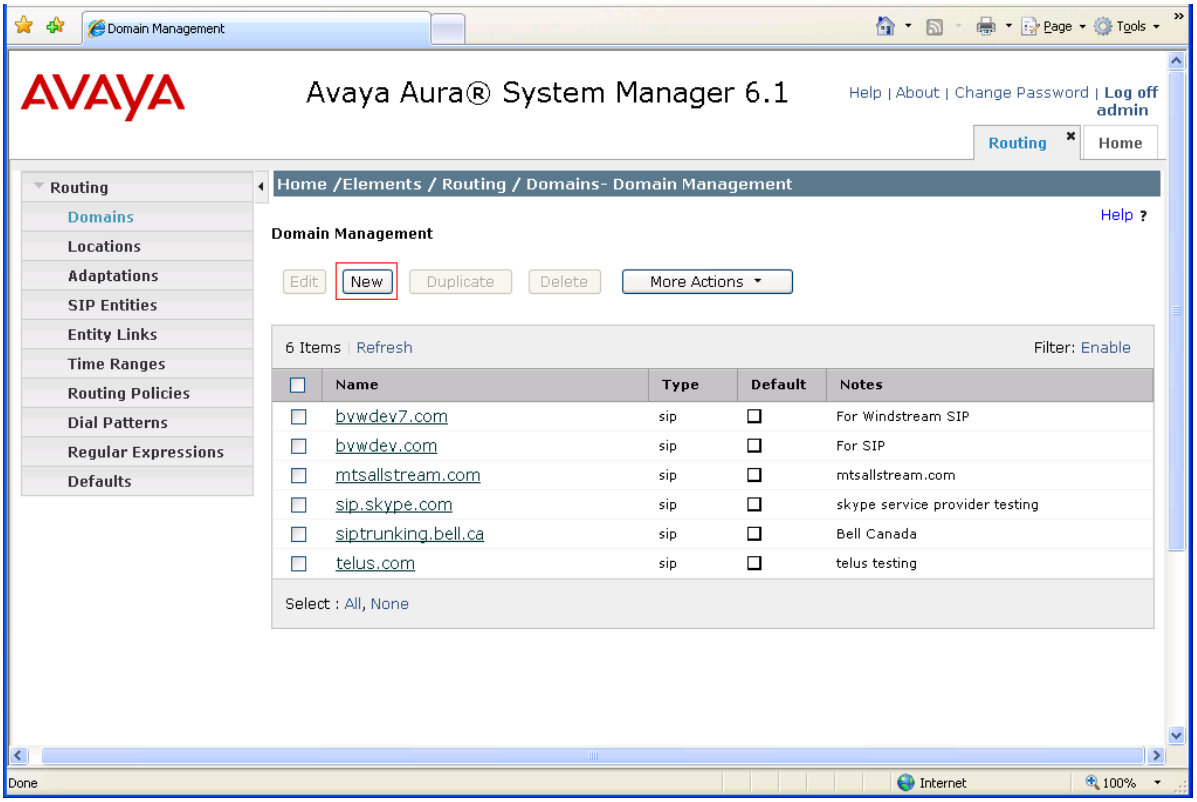

Figure 5: Domains – Domain Management page

Click on the **New** button to create a new SIP domain, the table of new domain appears, enter the **Name** **bvwdev.com**, **Type** as **SIP**, **Default** checkbox unchecked and **Notes** as shown in **Figure 6**. Click on the **Commit** button to finish adding the new sip domain **bvwdev.com**.

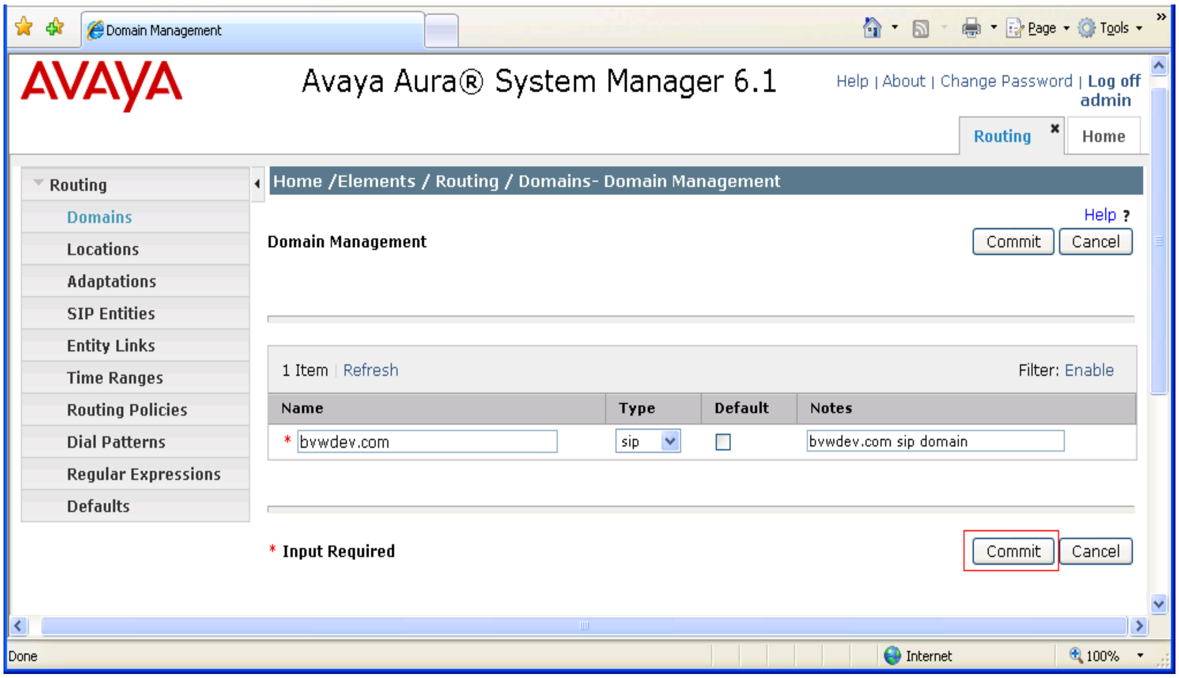

Figure 6: Add a new SIP domain in the Session Manager

5.2. Configure Locations

To configure a location in the Session Manager, in **Figure 4** of **Routing** page click on the **Locations** link and the **Locations Detail** appears in right-hand side, enter *Belleville* in the **Name** field, select the **Managed Bandwidth Units** as Kbit/sec, **Total Bandwidth** *1000000* (note that this is sample for this testing, bandwidth can be assigned based on individual network), **Default Audio Bandwidth** *100* Kbit/Sec as shown in **Figure 7** and click on the **Commit** button to finish creating the new location.

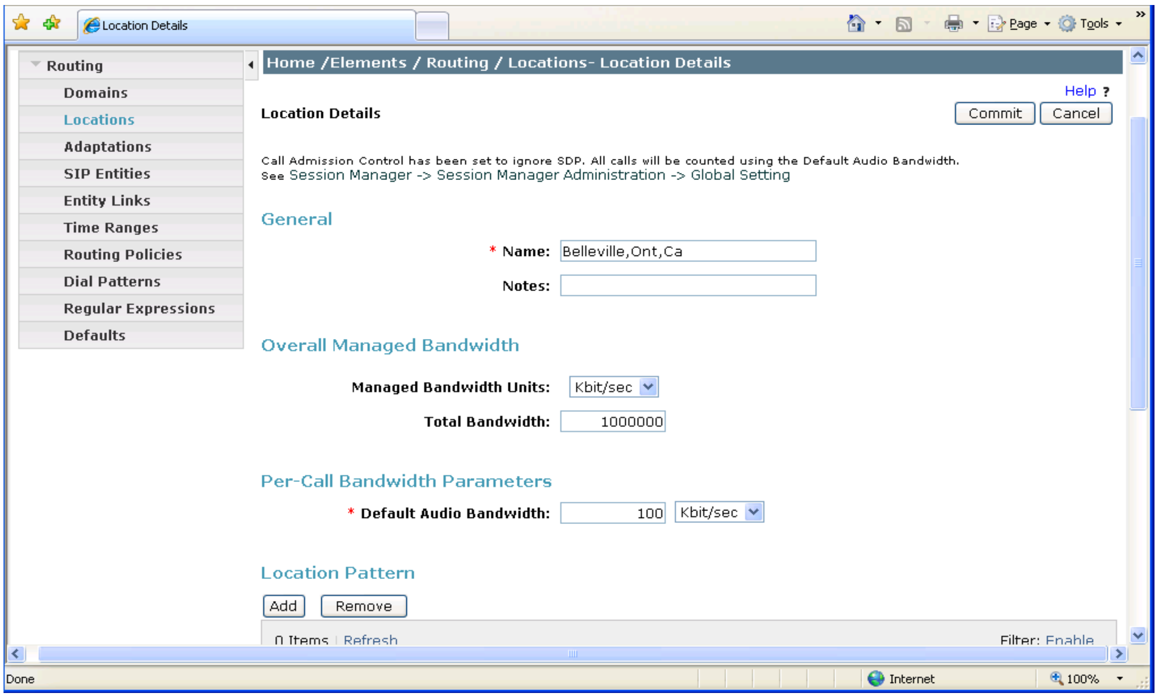

Figure 7: Adding a new location

5.3. Configure SIP Entities

To configure SIP Entities for CS1000 and RMX 4000, on left-hand side of **Routing** page click on the **SIP Entities** tab, the **SIP Entities** section appears in right-hand side of the **Routing** page (not shown), click on the **New** button to create a new SIP Entity for the CS1000 SIP Gateway, the **SIP Entity Details** section appears as shown in **Figure 8**, enter *cppm1* in the **Name** box, **FQDN or IP address** *135.10.97.130*, **Type** *Other*, **Locations** *Belleville* as defined in the **Section 5.2** and keep other values as default.

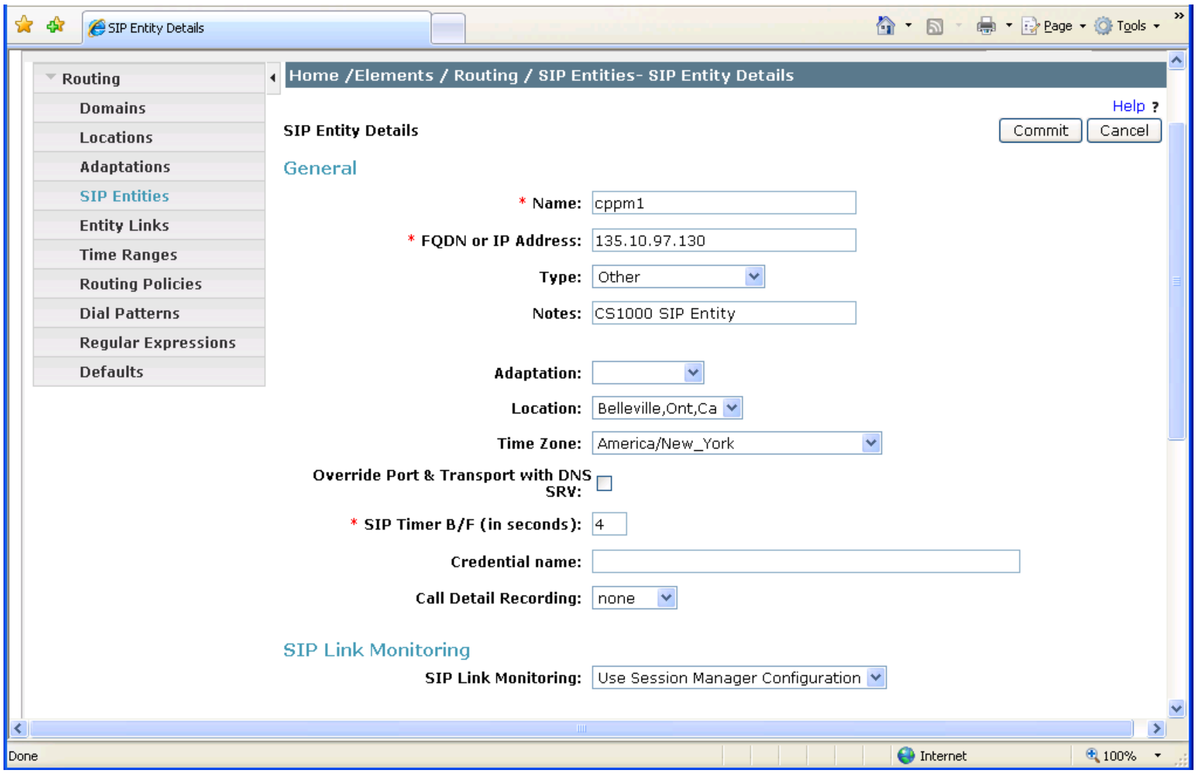

Figure 8: Adding new SIP Entity for CS1000

Repeat the same procedure to create a new SIP Entity for the Polycom RMX 4000 as shown in **Figure 9**.

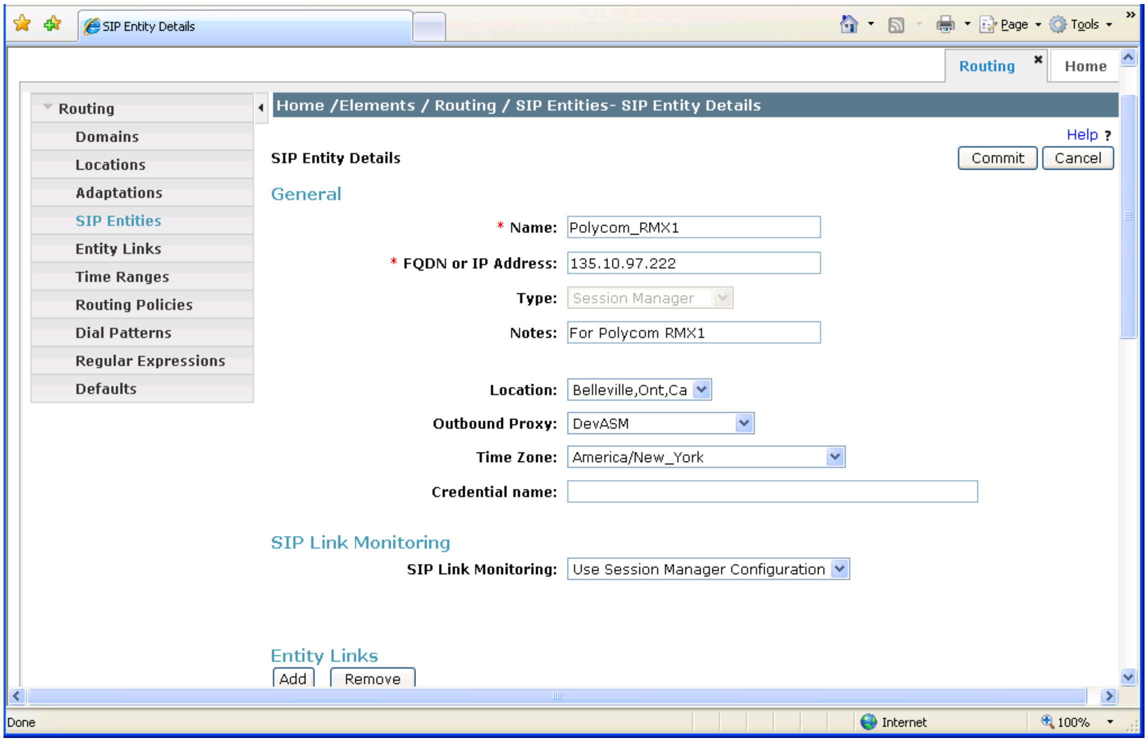

Figure 9: Adding new SIP Entity for Polycom RMX 4000

5.4. Configure Entity Links

To configure Entity Links, on left-hand side of **Routing** page click on the **Entity Links** tab, the **Entity Links** section appears in right-hand side of the **Routing** page (not shown), click on the **New** button to create a new entity link for the CS1000 SIP Entity above, the **Entity Links** section appears, enter *TCP_Link* in the **Name** box, select *DevASM* in the **SIP Entity 1**, **Protocol** *TCP*, **Port** *5060*, **SIP Entity 2** *cppm1* as defined in the **Section 5.3** and **Port** *5060* as shown in **Figure 10**.

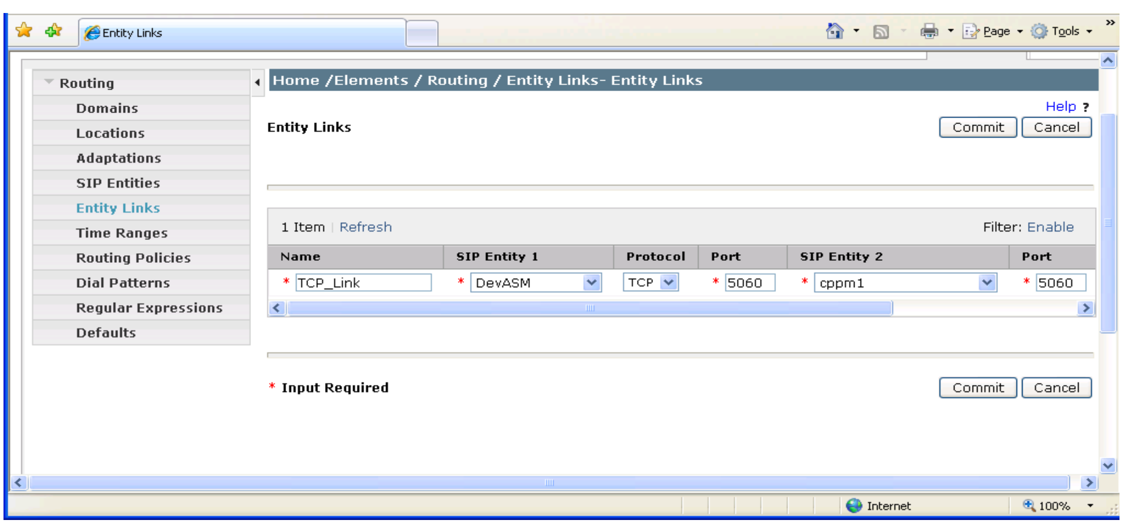

Figure 10: Adding new Entity Link for CS1000 SIP Entity

Repeat the same procedure for creating a new Entity Link for the Polycom RMX 4000 SIP Entity as shown in **Figure 11**.

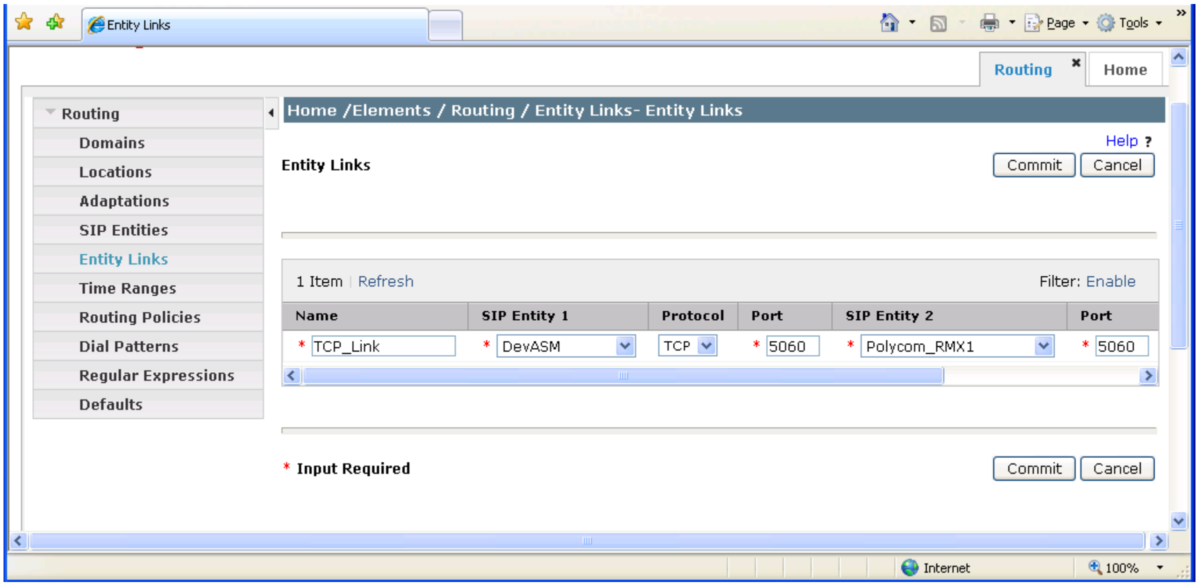

Figure 11: Adding new entity link for RMX 4000 SIP Entity

5.5. Configure Routing Policies

To configure a Routing Policy, on left-hand side of **Routing** page, click on the **Routing Policies** tab, the **Routing Policies** section appears in right-hand side of the **Routing** page (not shown), click on the **New** button to create a new route for the CS1000 SIP Entity, the **Routing Policy Details** section appears as shown in **Figure 12**. Enter *Route_2_CS1K* in the **Name** box, in the **SIP Entity as Destination** section click on the **Select** button and select the SIP Entity as *cppm1* (not shown) as defined in **Section 5.3**. Click on the **Commit** button to save the new route.

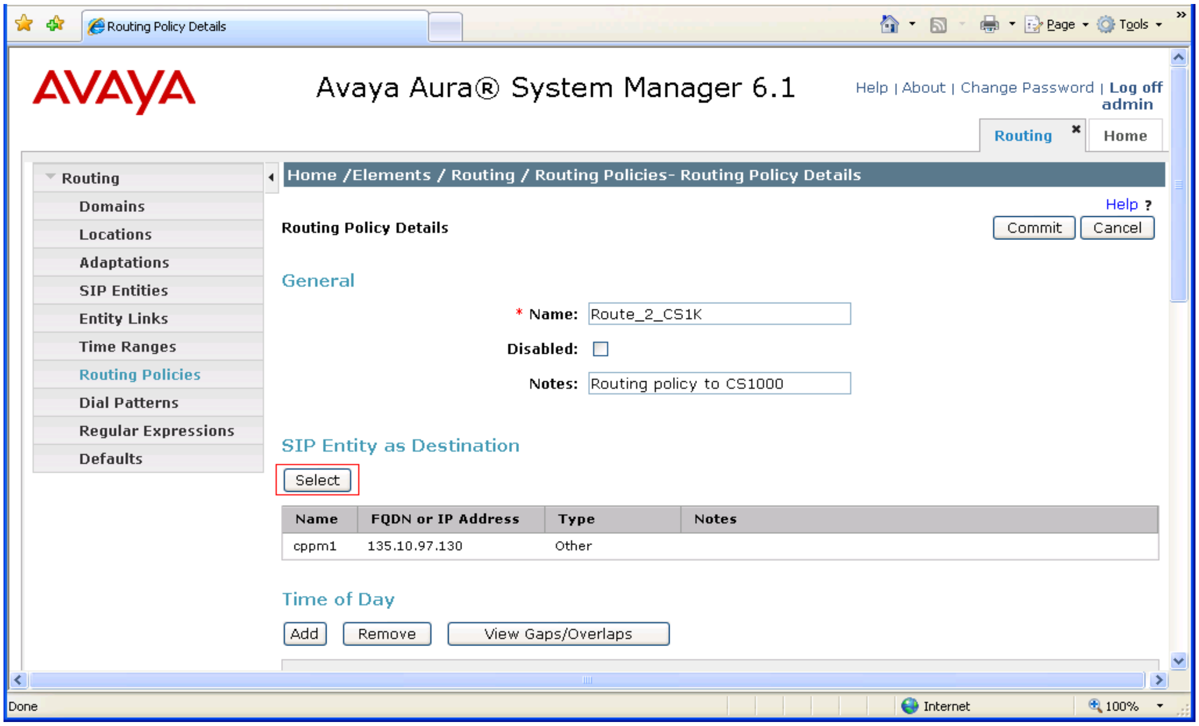

Figure 12: Adding new route for CS1000 SIP Entity

Repeat the same procedure above to create a new routing policy for the Polycom RMX 4000 SIP Entity e as shown in **Figure 13**.

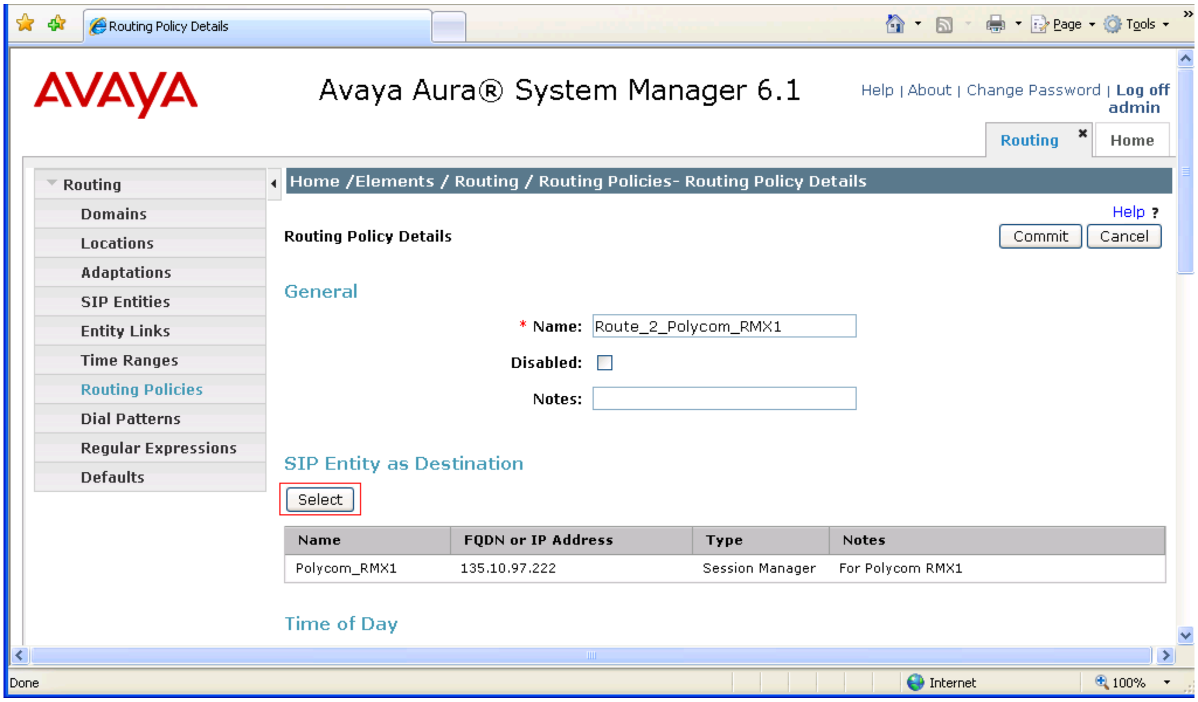

Figure 13: Adding new route for RMX 4000 SIP Entity

5.6. Configure Dial Patterns

To configure a Dial Pattern, on left-hand side of **Routing** page, click on the **Dial Patterns** tab, the **Dial Pattern** section appears in right-hand side (not shown), click on the **New** button to create a new dial pattern for the CS1000 routing, the **Dial Pattern Details** section appears as shown in **Figure 14**. In the **General** section, enter prefix *58* in the **Pattern** box, **Min** *5*, **Max** *5* (Because the length of directory number is 5), **SIP Domain** *bvwdev.com*. In the **Originating Locations and Routing Policies** section, click on the **Add** button to add the location *Belleville* as defined in the **Section 5.2** and **Routing Policy Name** *Route_2_CS1K* as defined in **Section 5.3**. Click on the **Commit** button to save the new dial pattern.

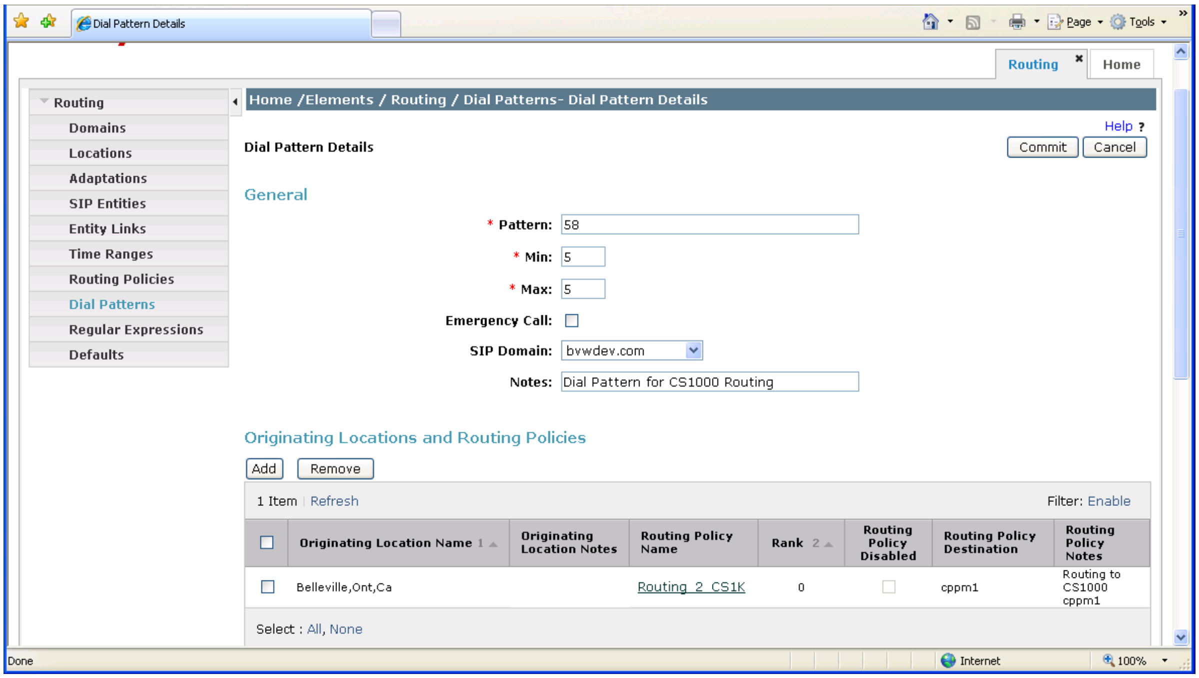

Figure 14: Adding new pattern for CS1000 Route

Repeat the same procedure above to create the new dial pattern for the Polycom RMX 4000 route as shown in **Figure 15**.

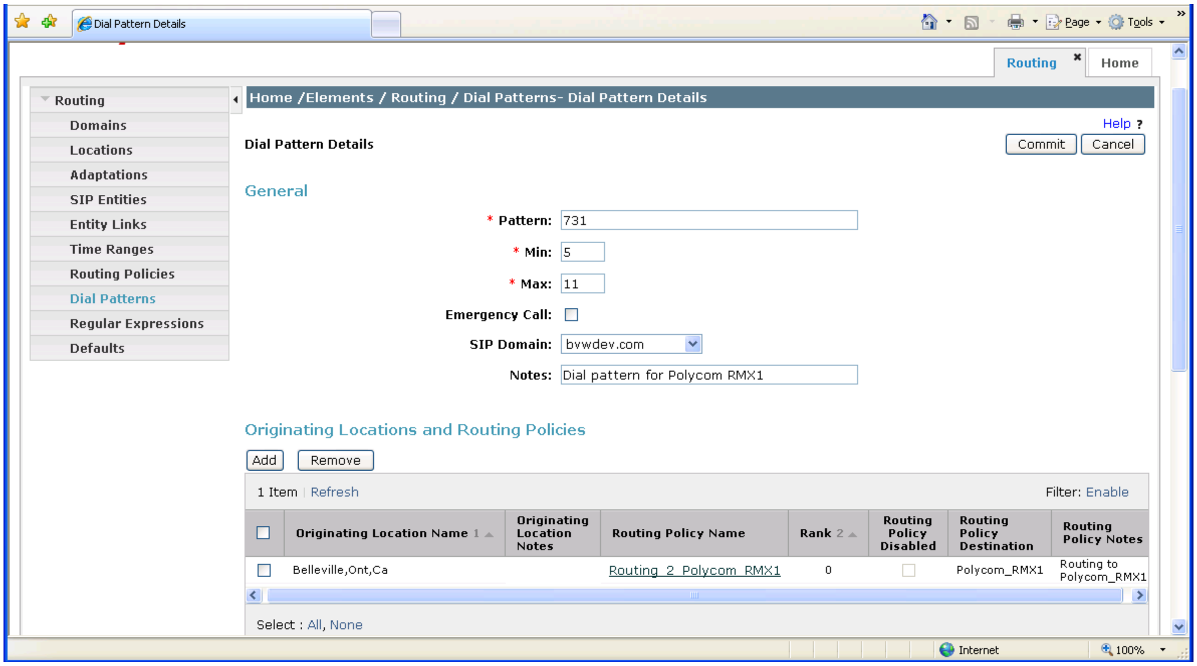

Figure 15: Adding new dial pattern for RMX 4000 Route

6. Configure Avaya Communication Server 1000

This document assumes that the Avaya Communication Sever 1000 system was properly installed and configured as per the product documents, this section provides the steps on how to provision the CS1000 working with the Session Manager and Polycom RMX 4000. For more information about how to install and configure Communication Sever 1000, please refer to **Section 11 [3-6]**.

The following summarizes the tasks that need to be done in the CS 1000 system:

- Register the CS1000 SIP Signaling Gateway to Session Manager.
- Configure D-Channel for SIP Trunk
- Configure Zone Bandwidth
- Configure SIP Route.
- Configure SIP Trunks.
- Configure CDP Dialing plan.

6.1. Register the CS1000 SIP Signaling Gateway to Session Manager

To register the CS1000 SIP Signaling Gateway to the Session Manager, follow the procedures below: Log in to the Unified Communication Management (UCM) managing the CS1000 system that needs to be configured, the UCM login window is shown as **Figure 16**.

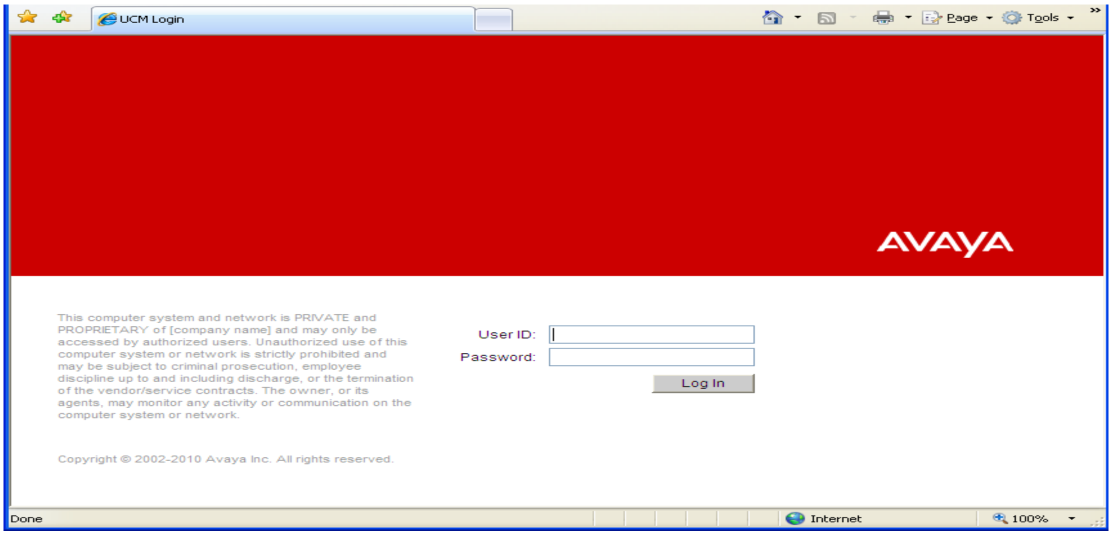

Figure 16: UCM Login window

Enter the username "admin" and its password in the **User ID** and **Password** boxes and click on the **Login** button. The homepage of the UCM appears as shown in **Figure 17**.
Click on the **Element Name** link (in this sample it is *EM on cppm1*) that manages the CS1000 system,.

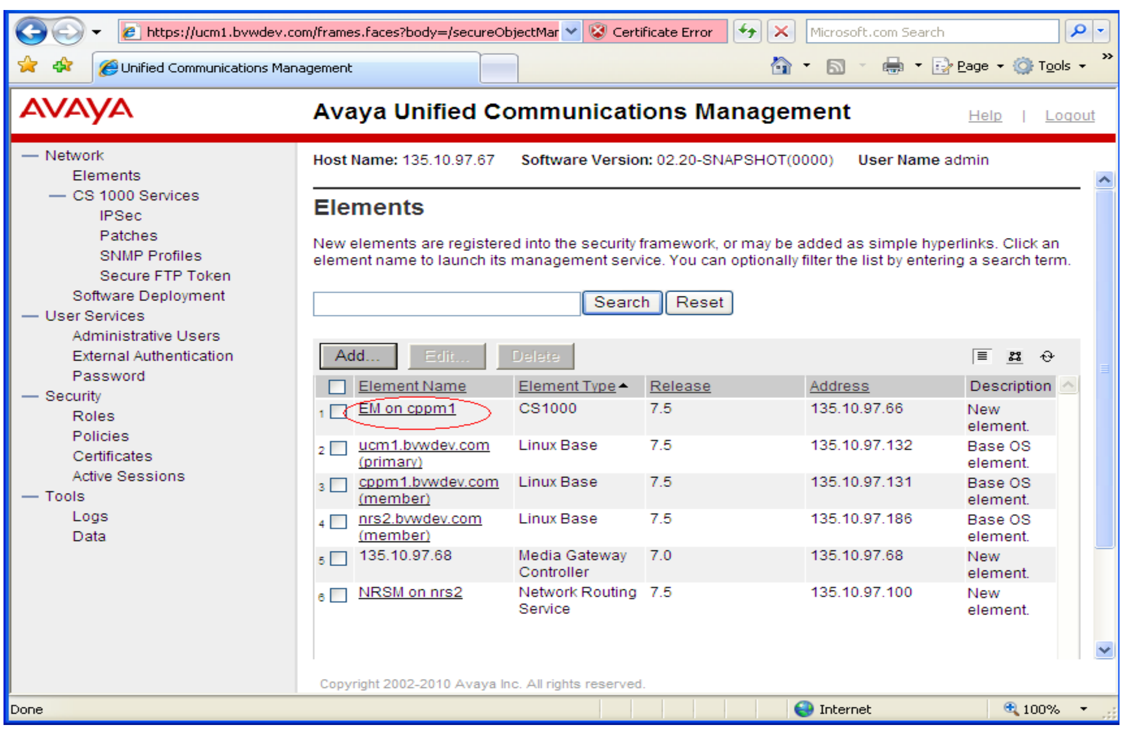

Figure 17: UCM Home Page

The Element Manager window appears as shown in **Figure 18**.

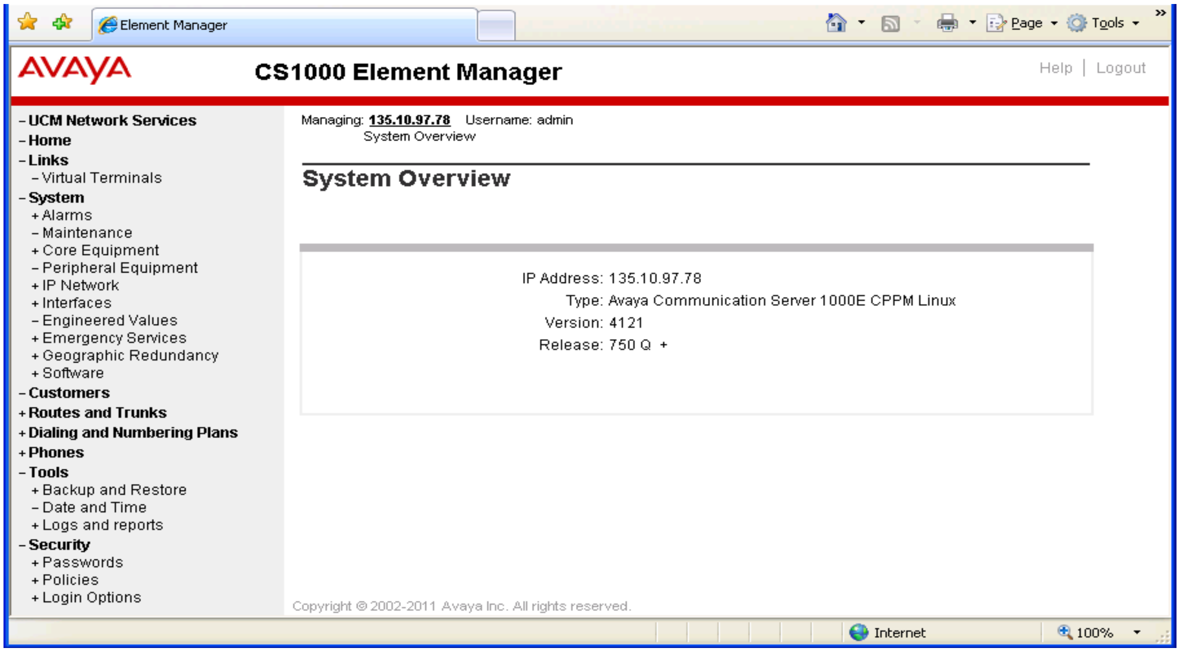

Figure 18: CS1000 Element Manager Home page

On left-hand side of the Element Manager window and under the **System** tab, expand **IP Network** > **Nodes: Servers and Media Cards**, the **IP Telephony Nodes** screen is displayed in the right-hand side of the window as shown in **Figure 19**. Click on the **Node ID** (in this sample it is **551**) which has the **SIPGw** application enabled.

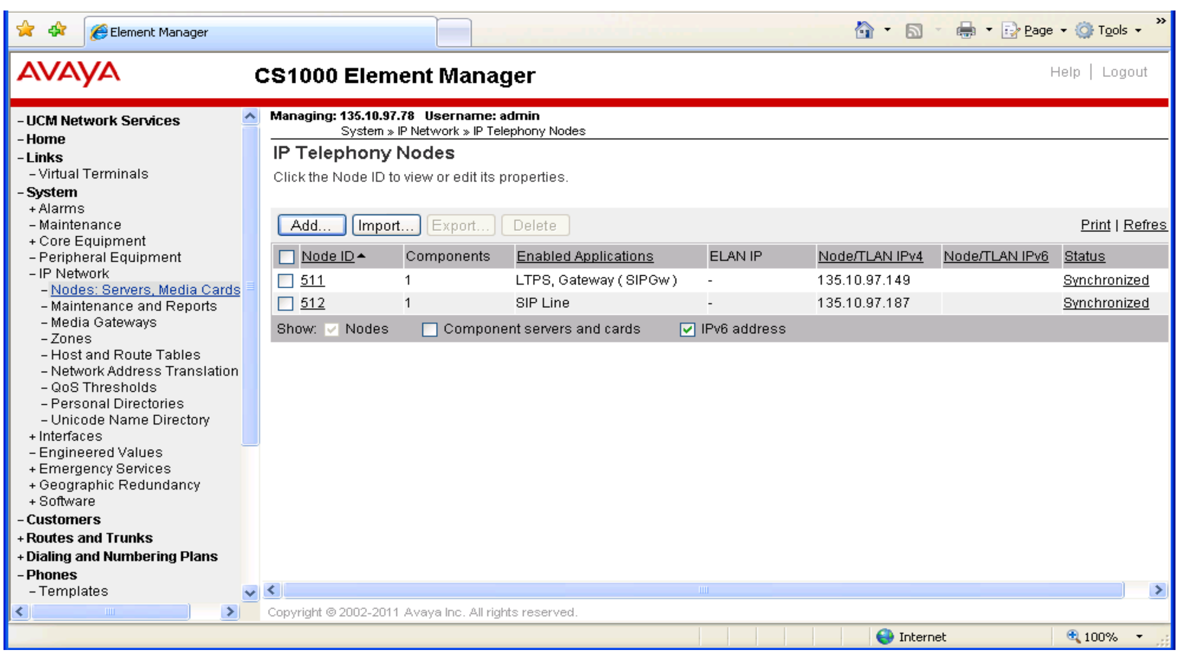

Figure 19: IP Telephony Nodes Page

The Node **511** detail appears as shown in **Figure 20**. Under **Applications**, click on the **Gateway (SIPGw)** application link.

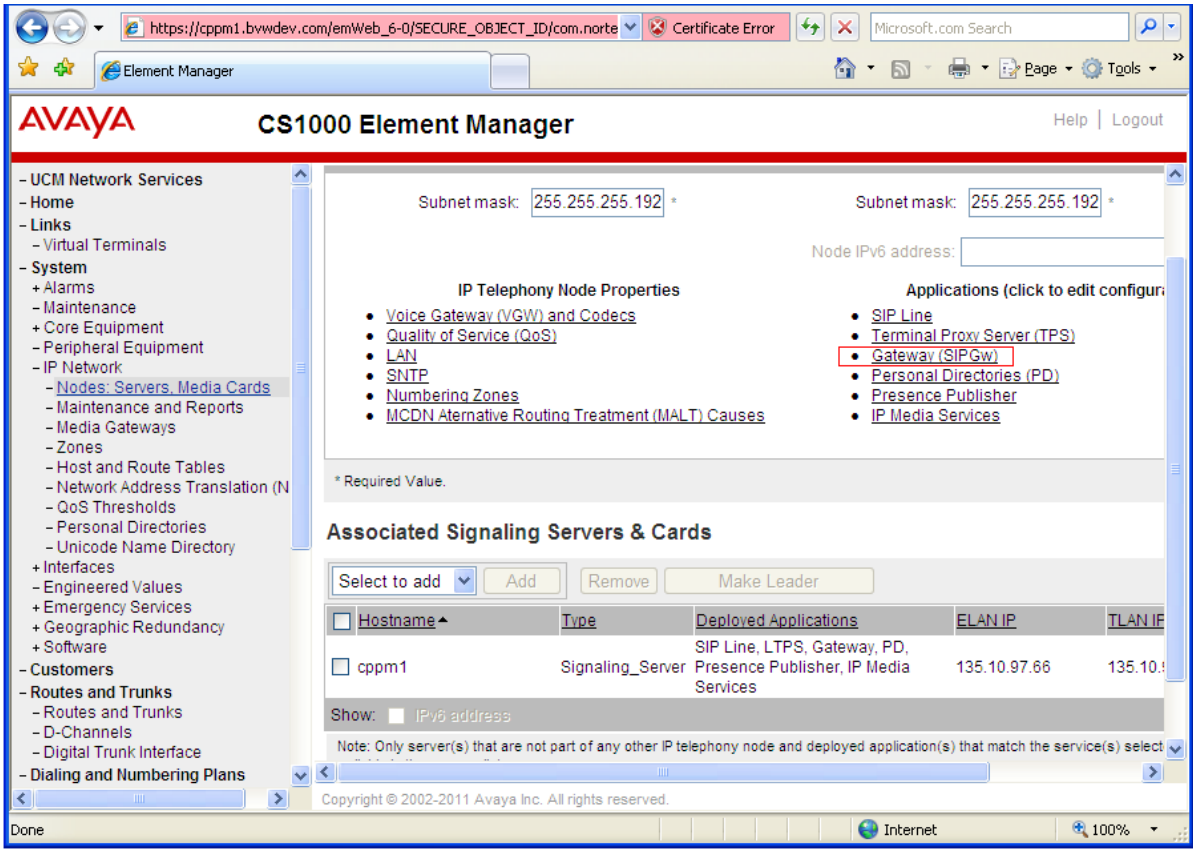

Figure 20: IP Telephony Node Detail Page

The **Node ID: 551 – Virtual Trunk Gateway Configuration Details** appears. In the **General** section, enter the domain *bvwdev.com* in the **SIP domain name** box as defined in the Section 5, and **Local SIP Port** *5060*, **Gateway endpoint name** *cppm1*, and **Application node ID** as **551** as shown in **Figure 21.**

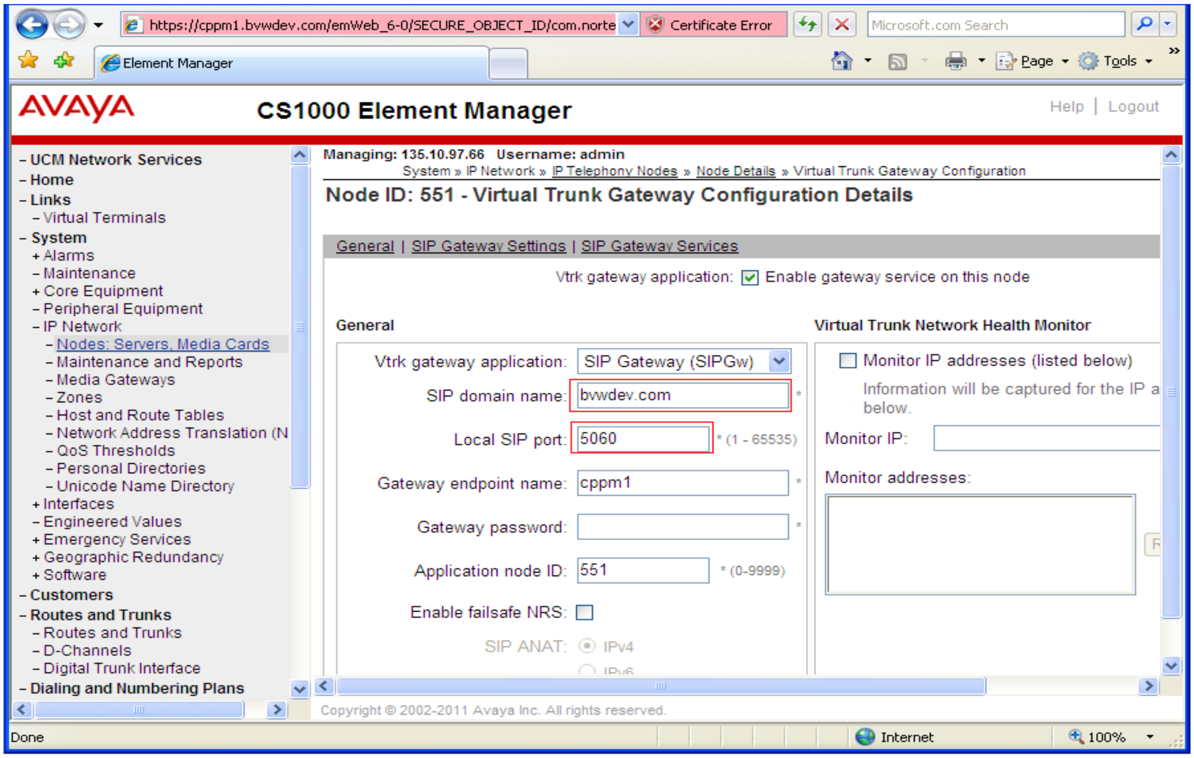

Figure 21: Node ID: 511 – Virtual Trunk Gateway Configuration Details

Continue scrolling down to the section **SIP Gateway Settings (not shown in the Figure)**, in the **Proxy Or Redirect Server** section, enter the IP address of the Session Manager signaling interface in the field **Primary TLAN IP address**, **Port** *5060*, **Transport protocol** *UDP* as shown in **Figure 22**.

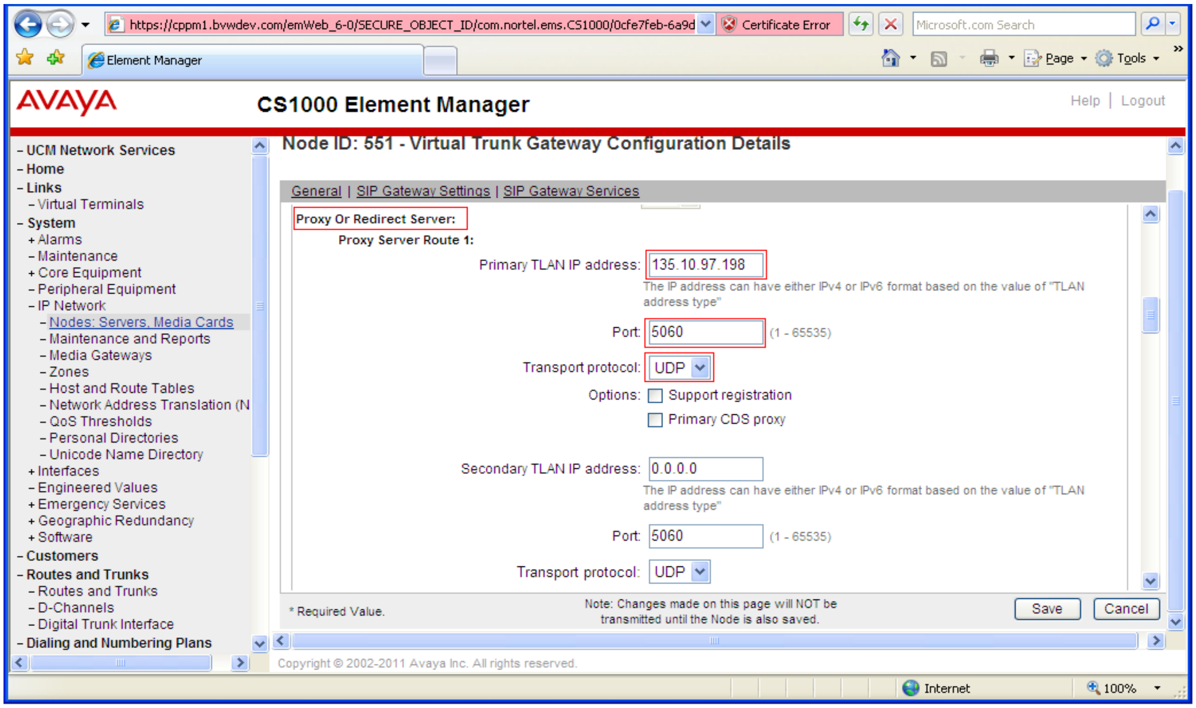

Figure 22: The SIP Gateway Setting of Telephony Node

Click on the **Save** button at the bottom of this page to save the changes in the Node ID 551, the **Node ID 511 - Virtual Trunk Gateway Configuration Detail** window will be closed. Back in the **Node ID: 511** page**,** click on the **Save** button (not shown), the **Node Saved** window appears as shown in **Figure 23**. Click on the **Transfer Now…** button..

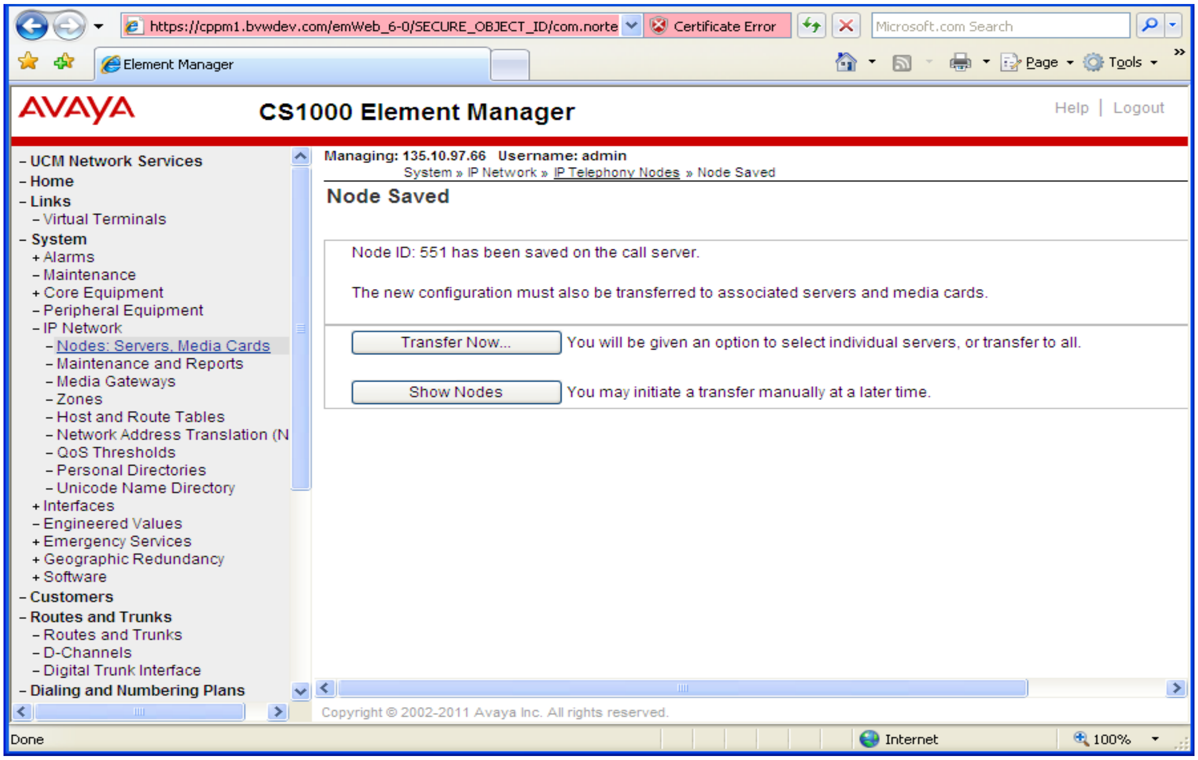

Figure 23: Node Saved Page

The **Synchronize Configuration Files (Node ID)** page appears as shown in **Figure 24**, click on the associated server (in this sample it is *cppm1*) and click on the **Start Sync** button to start transferring the changes to this server

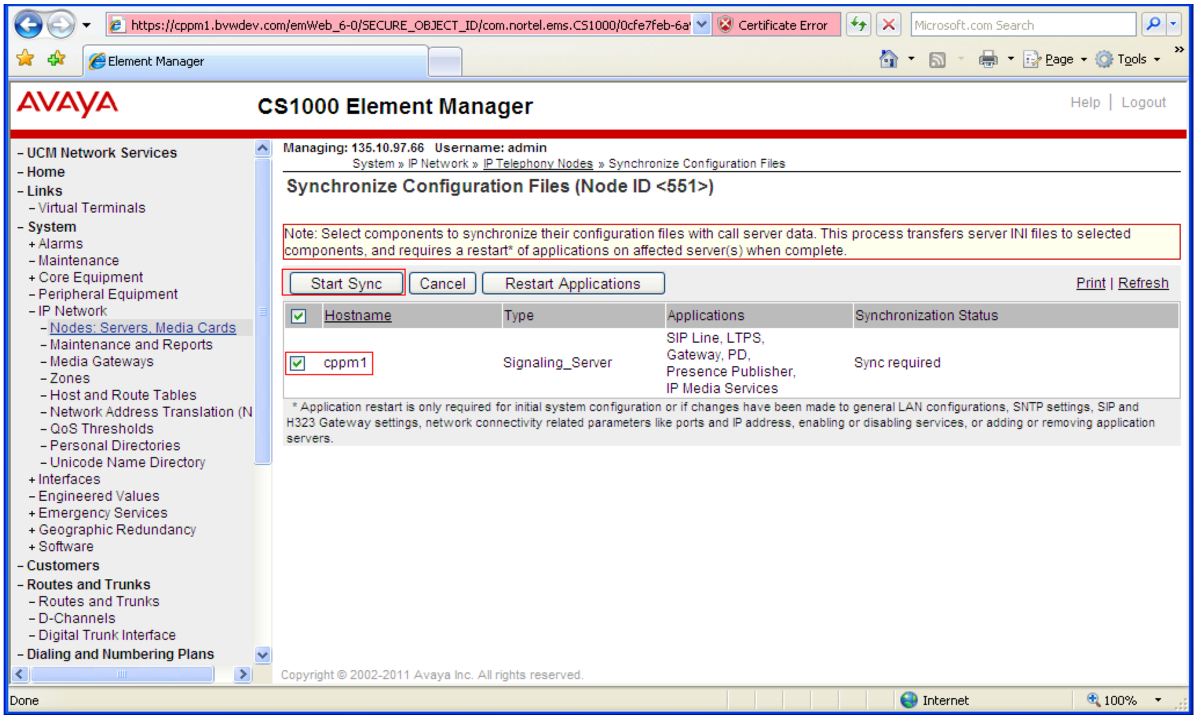

Figure 24: Synchronize Configuration Files (Node ID <551>) page

6.2. Configure D-Channel for SIP Trunk

To configure a D-Channel for a SIP trunk in the Element Manager, from the homepage of Element Manager, expand the menu **Routes and Trunks > D-Channels** and select the **D-Channels** tab, the **D-Channel** page appears in the right-hand side as shown in **Figure 25**. In the **Configuration** section of this page, select an available D-Channel in the **Choose a D-Channel Number** dropdown list, select the type of D-Channel as *DCH* and click on the to **Add** button.

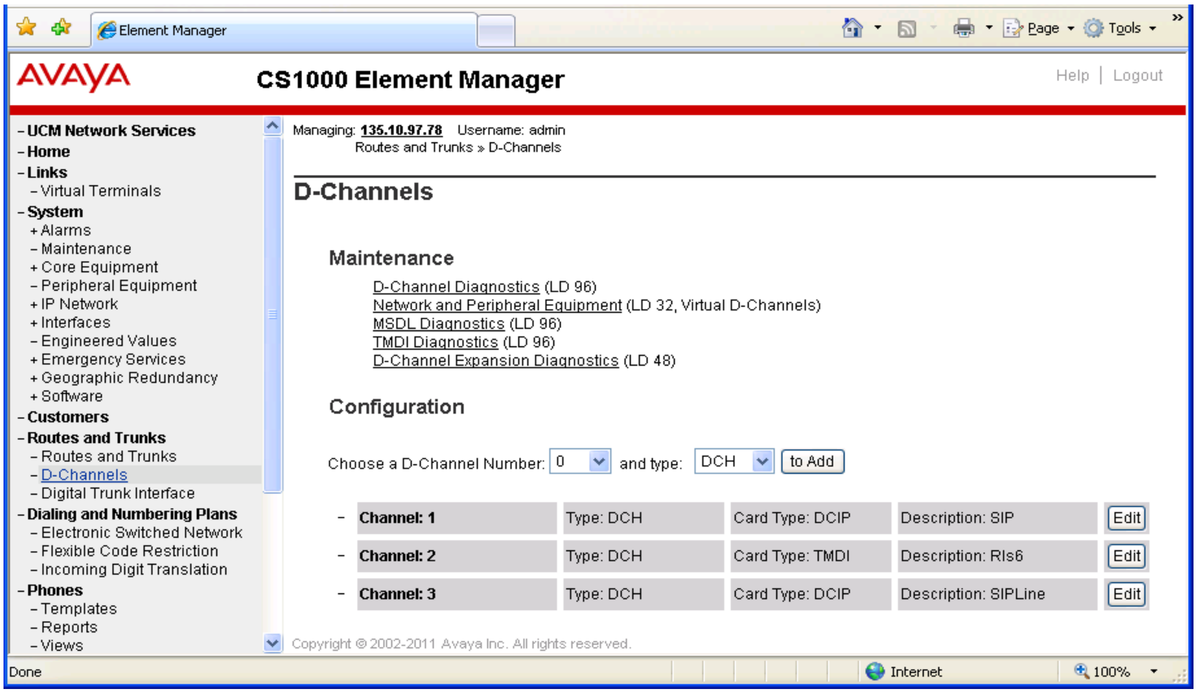

Figure 25: D-Channels page

The **Basic Configuration** of new D-Channel appears as shown in **Figure 26**. Select *D-Channel is over IP (DCIP)* in the **D-Channel Card Type** field, enter a description in the **Designator** box and keep other values as default.

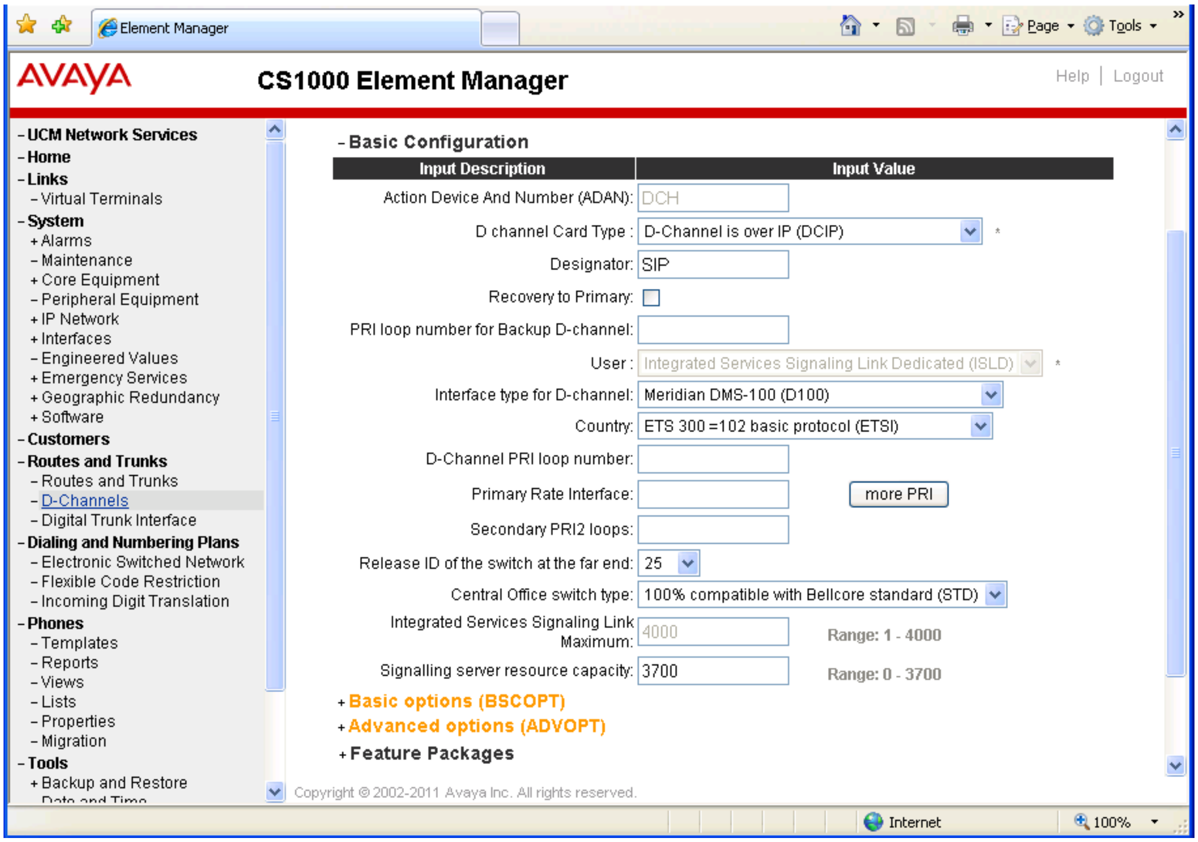

Figure 26: Basic Configuration section of D-Channel

Continue expanding the **Basic options (BSCOPT)** subsection of the **Basic Configuration** Section, in this section click on **Edit** button of the **Remote Capabilities** field, the **Remote Capabilities Configuration** page appears as shown in **Figure 27**. Check on the checkbox of **Network name displayed method 2 (ND2)** and click on the **Return – Remotes Capabilities** button to go back to the **Basic options** section. Keep other values of this section as default.

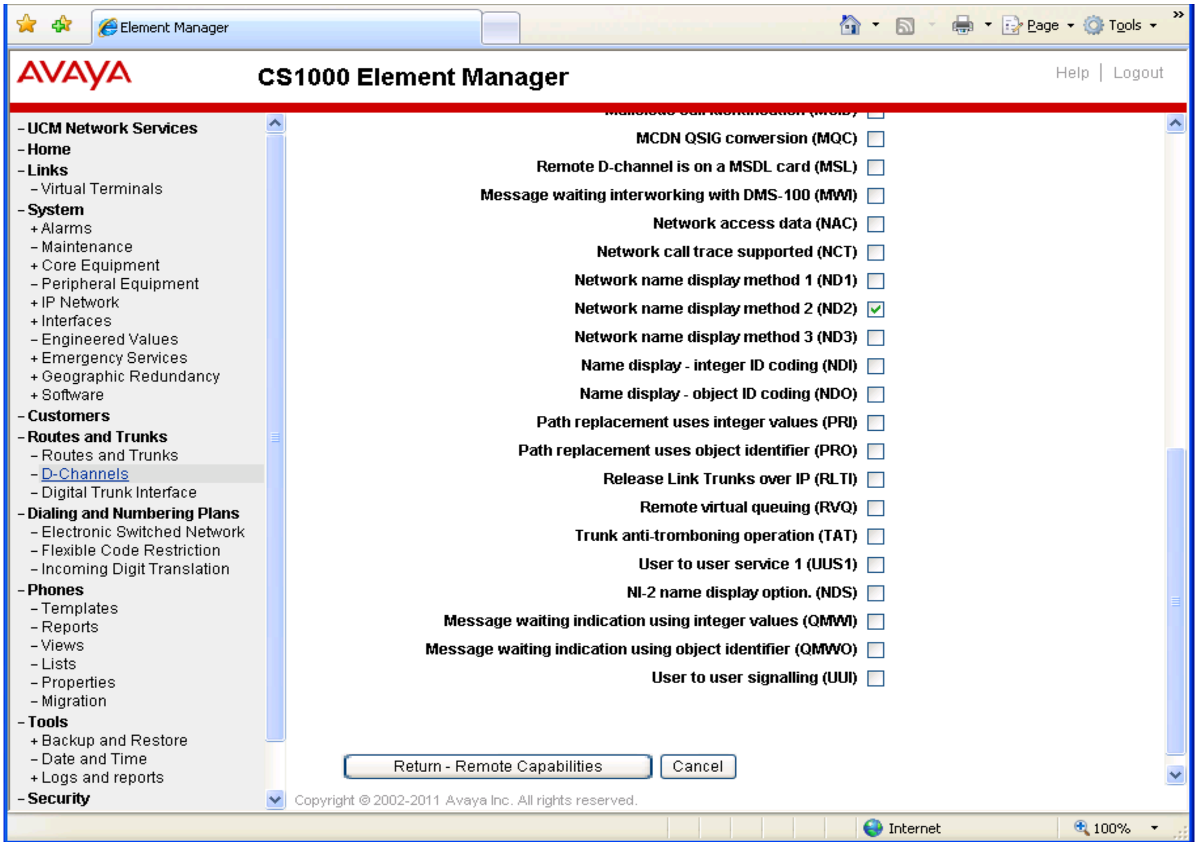

Figure 27: Remote Capability page

Continue expanding the **Advanced options (ADVOPT)** subsection of the **Basic Configuration** section, the **Advanced options** section appears as shown in **Figure 28** below. Leave all the values at default.

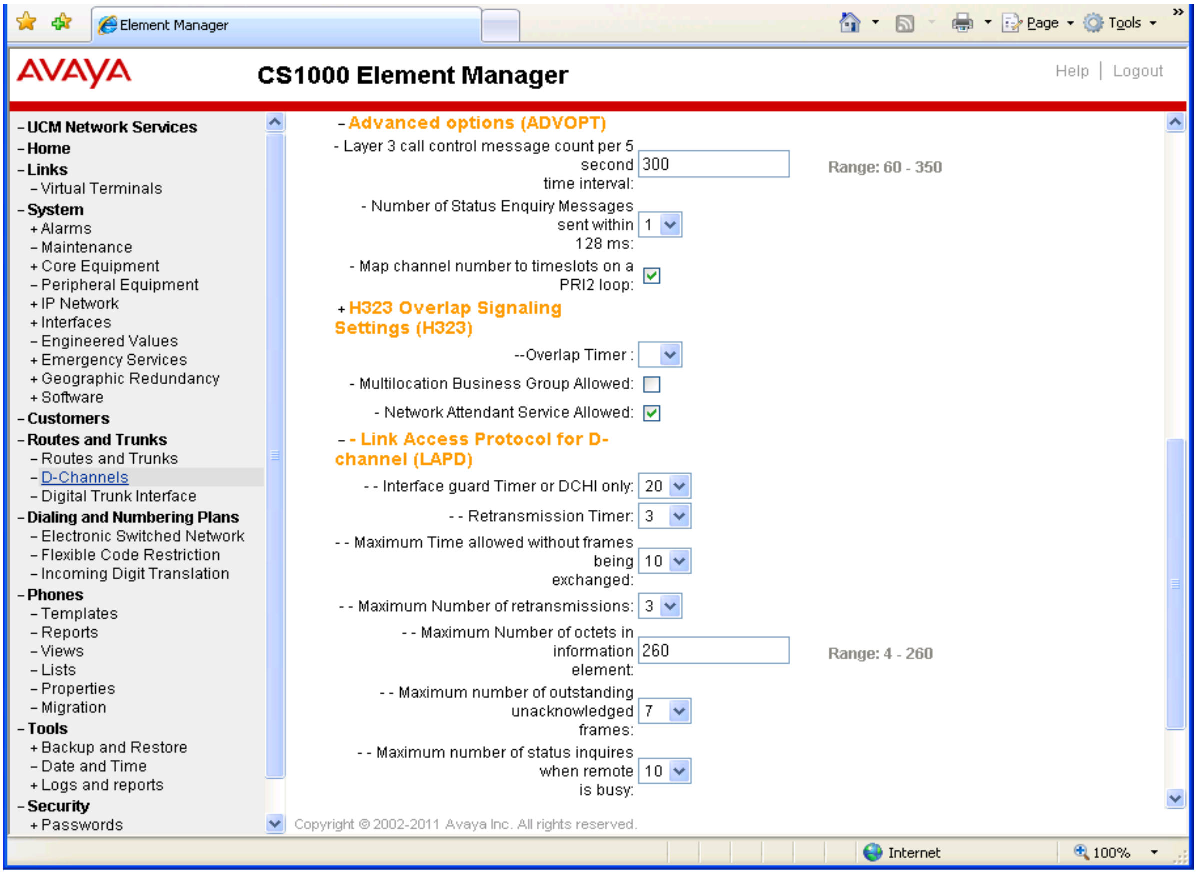

Figure 28: Advanced options (ADVOPT)

Keep the **Feature packages** section of the new D-Channel page as default and finally click on **Submit** button in the bottom of the new **D-Channel** page to complete adding the new D-Channel (not shown)

6.3. Configure Zone Bandwidth

To configure a Zone in the Element Manager, from the homepage of Element Manager, expand the menu **System > IP Network > Zones** and select the **Zones** tab, the **Zones** section appears in the right-hand side as shown in **Figure 29**. Click on the **Bandwidth Zones** link, the **Bandwidth Zones** page appears (not shown) and then click on the **Add** button to add a new zone.

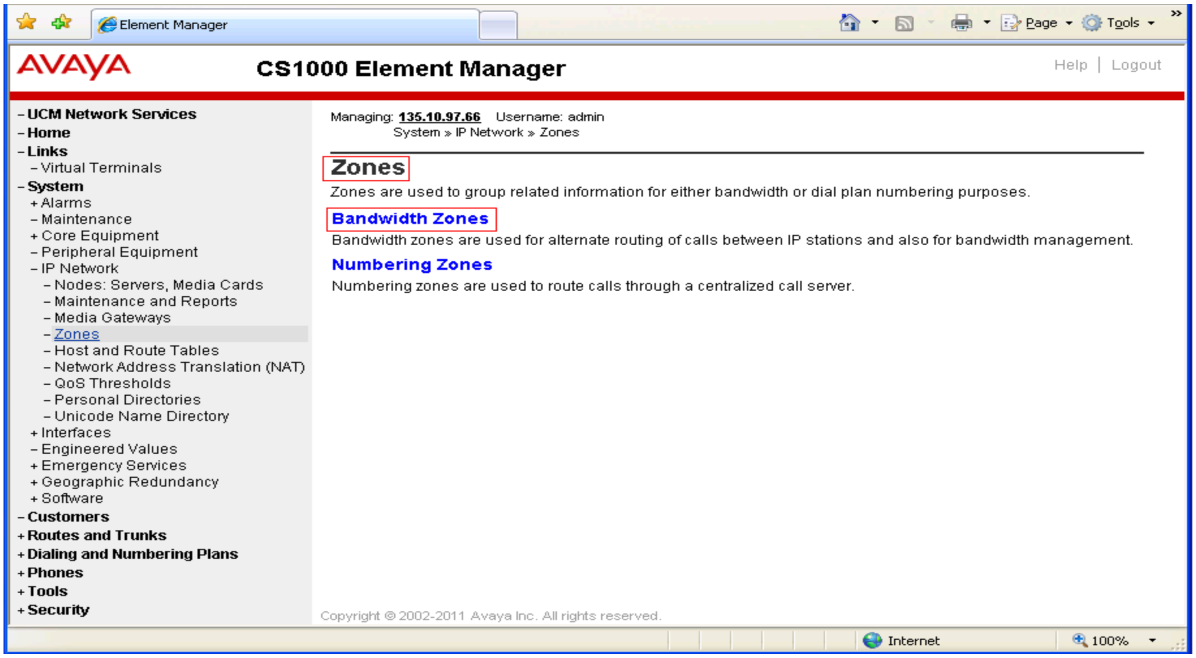

Figure 29: Zones Page

The **Zone Basic Property and Bandwidth Management** page appears as shown in **Figure 30**. Enter *4* in the **Zone Number**, select **Zone Intent (ZBRN)** as *VTRK* (because this zone is used for virtual trunks) and keep other fields as default. Click on the **Save** button to save changes.

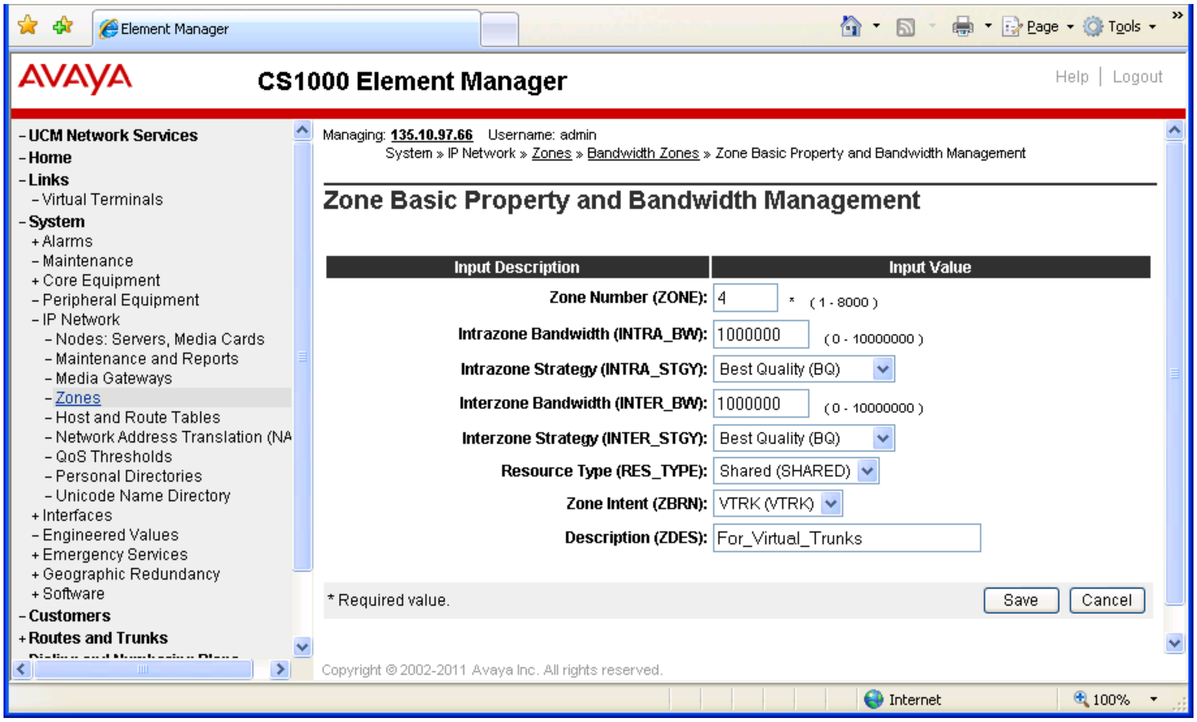

Figure 30: Zone Basic Property and Bandwidth Management

6.4. Configure SIP Route

To configure a SIP Route in the Element Manager, from the homepage of Element Manager navigate to menu **Routes and Trunks > Routes and Trunks**, the **Routes and Trunks** page appears in the right-hand side as shown in **Figure 31**.

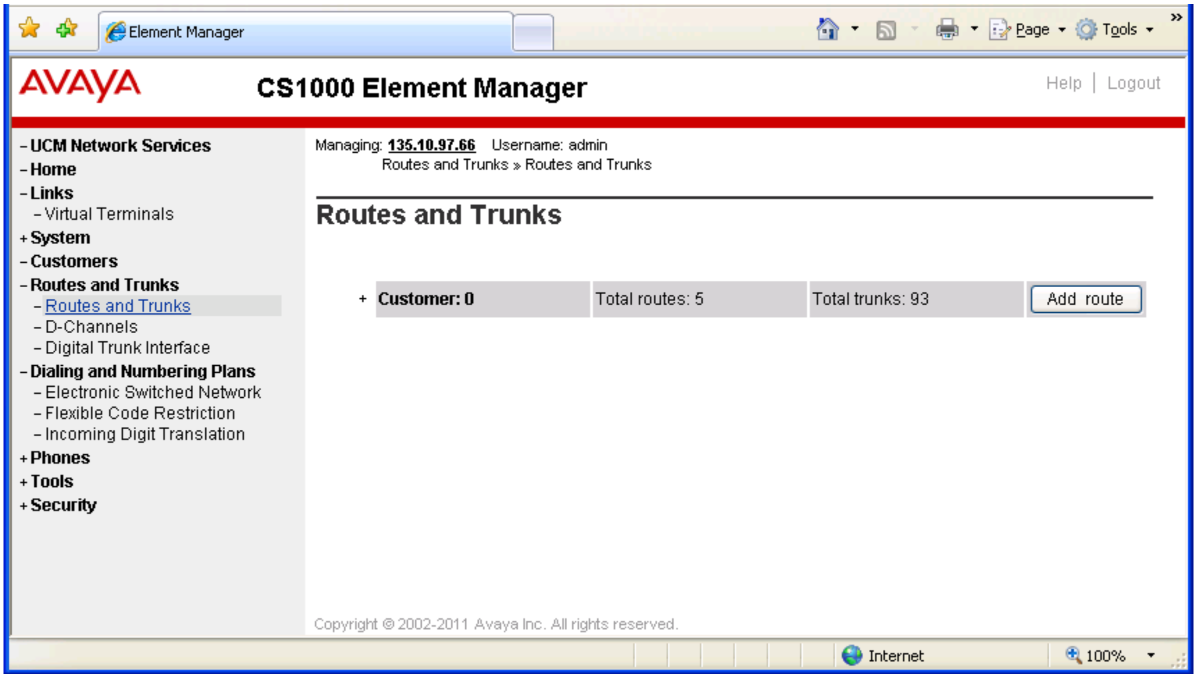

Figure 31: Routes and Trunks page

Identify which customer the new route is going to be added (in case there are many customers in the CS1000 system, in this sample there is just one, *Customer 0* so the route is going to be added for this customer) and then click on the **Add route** button of that customer, the **New Route Configuration** page appears as shown in **Figure 32** that consists of 5 sections: **Basic Configuration**, **Basic Route Options**, **Network Options**, **General Options**, and **Advanced Configurations**.

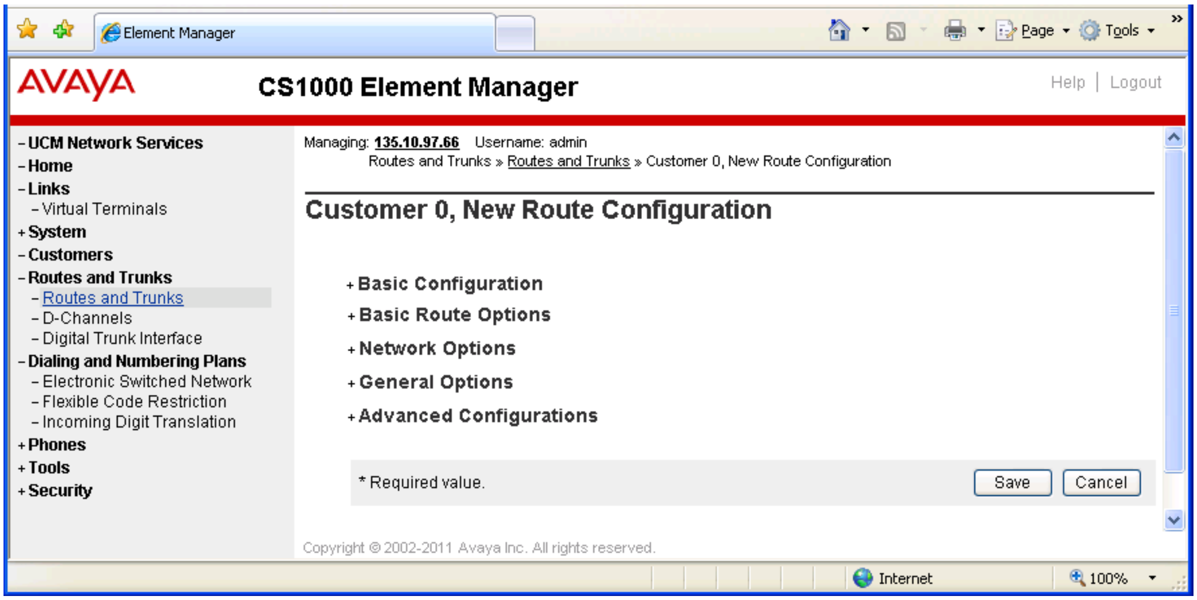

Figure 32: New Route Configuration page

Figure 33 below shows the **Basic Configuration** section with typical values entered for a SIP Route:

- **Route number (ROUT)**: *1*
- **Trunks type (TKTP)**: *TIE trunk data block(TIE)*
- **Incoming and outgoing trunk (ICOG)**: *Incoming and Outgoing (IAO)*
- **Access code for the trunk route (ACOD)**: *8001*
- **The route if for a virtual trunk route (VTRK)**: Checked
- **Zone for codec selection and bandwidth management (ZONE)**: *4* - as defined in the **Section 6.3**
- **Node ID of signaling server of this route (NODE)**: *551* - This Node is used to register to the Session Manager in the **Section 6.1**
- **Calling number dialing plan (CNDP)**: *Coordinated dialing plan (CDP)* - because the CDP dialing plan was used for this route.

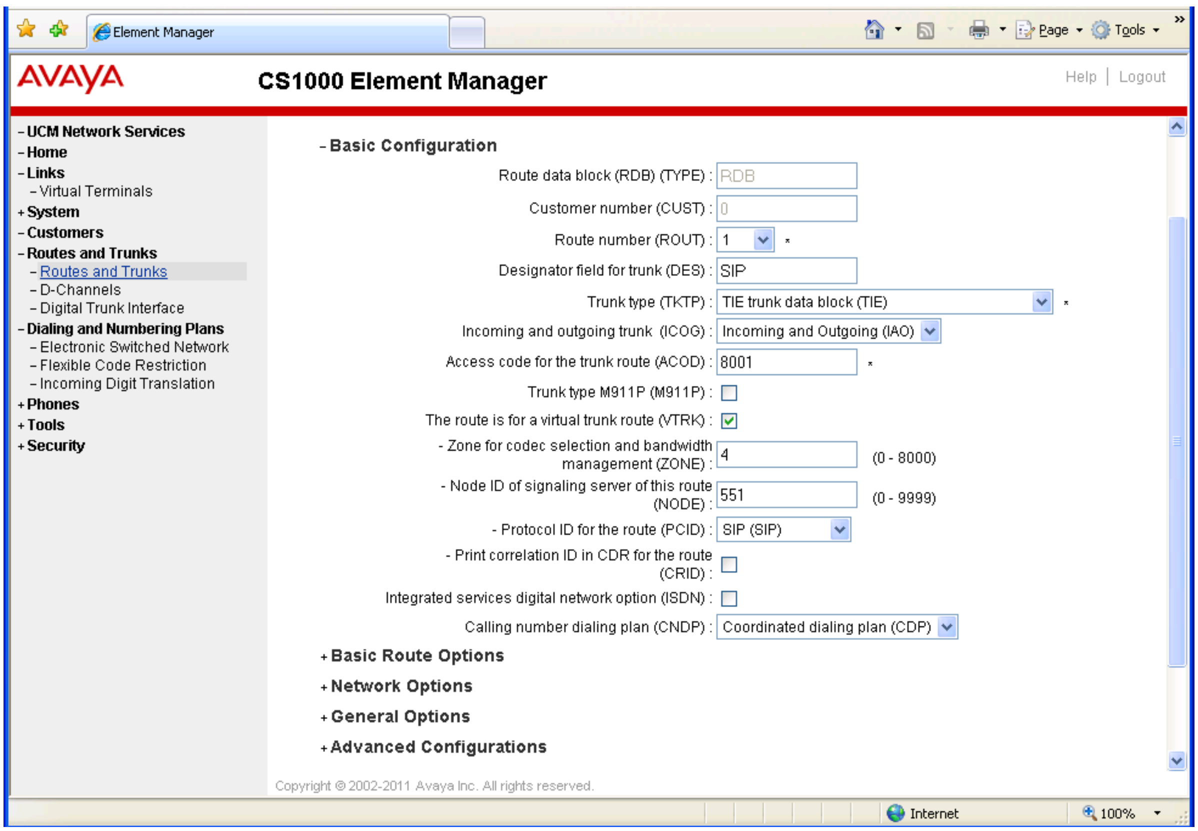

Figure 33: Basic Configuration section of new route

Check the **Integrated services digital network option (ISDN)** in the **Basic Configuration** section, **Figure 34** below shows the sub options for this feature enabled. The important values are entered as samples below.

- **Mode of operation (MODE)**: *Route uses ISDN Signaling Link (ISLD)*
- **D channel number (DCH)**: *10* - this is D-Channel for SIP Trunk as defined in the **Section 6.2**
- **Interface type for route (IFC)**: *Meridian 1 (SL1)*
- **Private Network Identifier (PNI)**: *1*
- **Network calling name allowed (NCNA)**: Checked.
- **Network call redirection (NCRD):** Checked

Keep other values as default, as shown in **Figure 34** below.

-

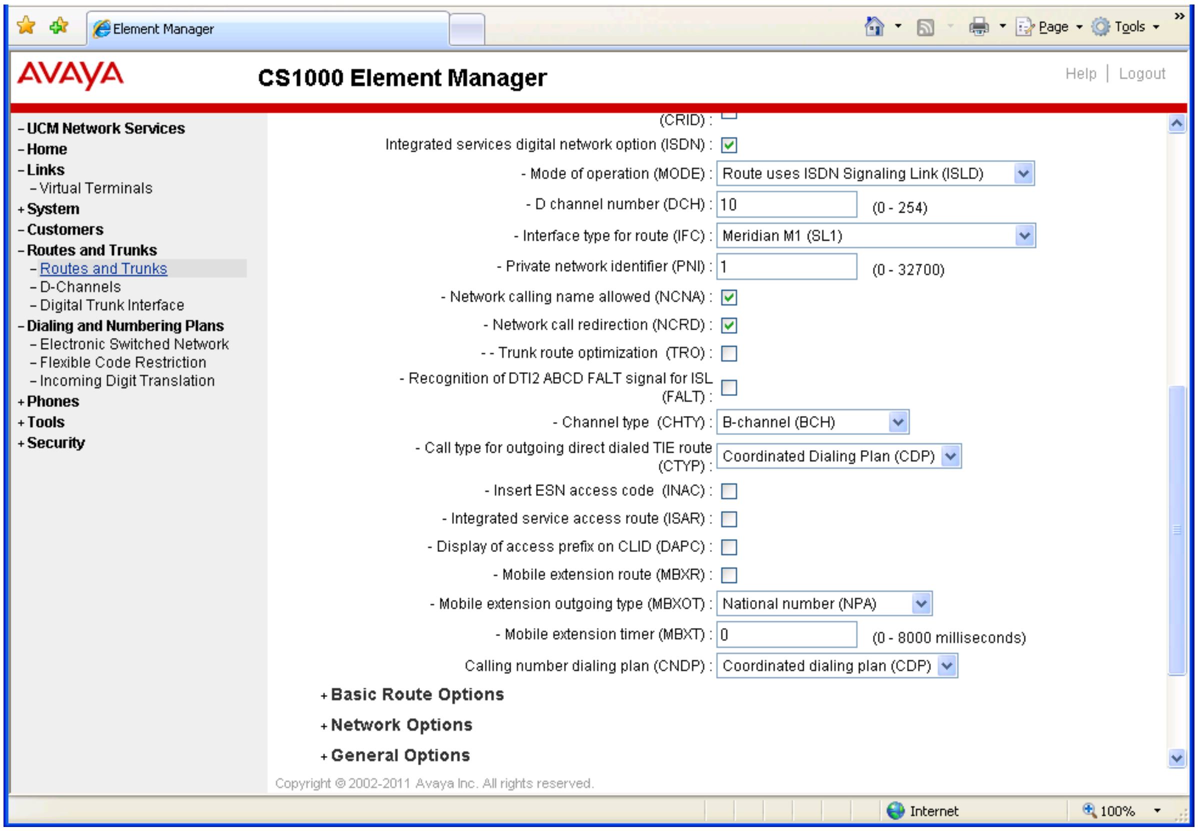

Figure 34: **Integrated services digital network option (ISDN) option page**

Continue expanding the **Basic Route Options** section and keep default values as shown in **Figure 35**.

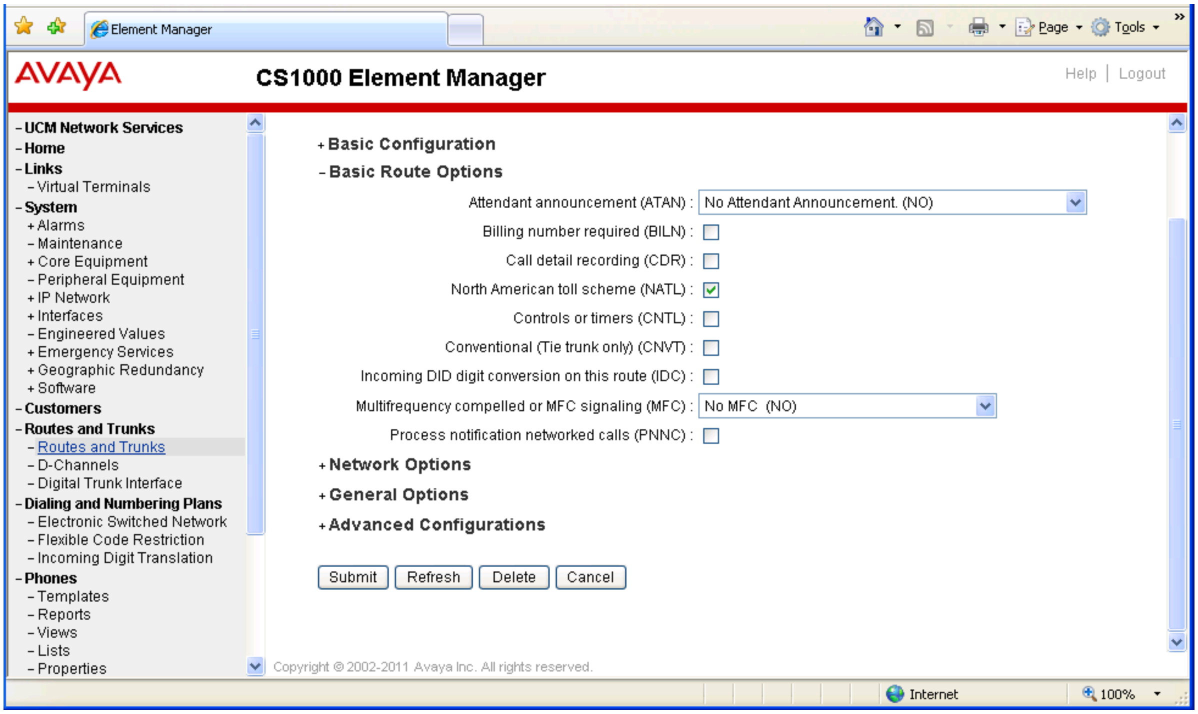

Figure 35: Basic Route Options of new Route

Continue expanding the **Network Options** and keep its default values as shown in **Figure 36**.

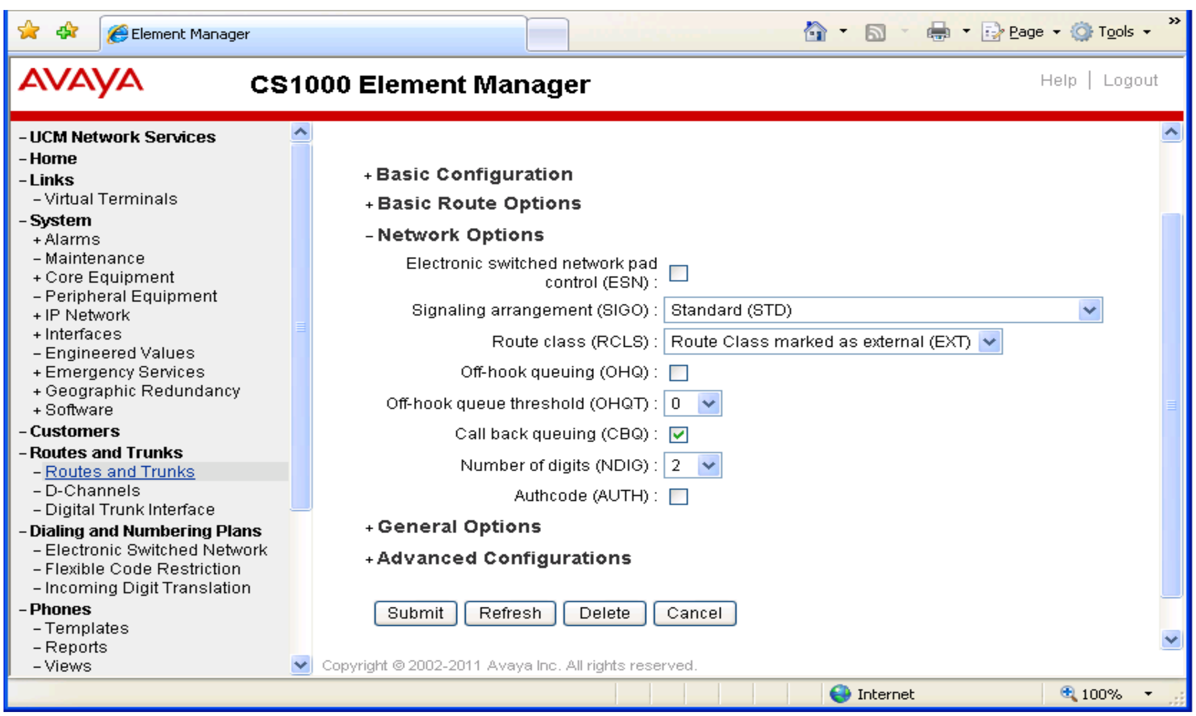

Figure 36: Network Options of new Route

Continue expanding the **General Options** section and keep its default values as shown in **Figure 37** below.

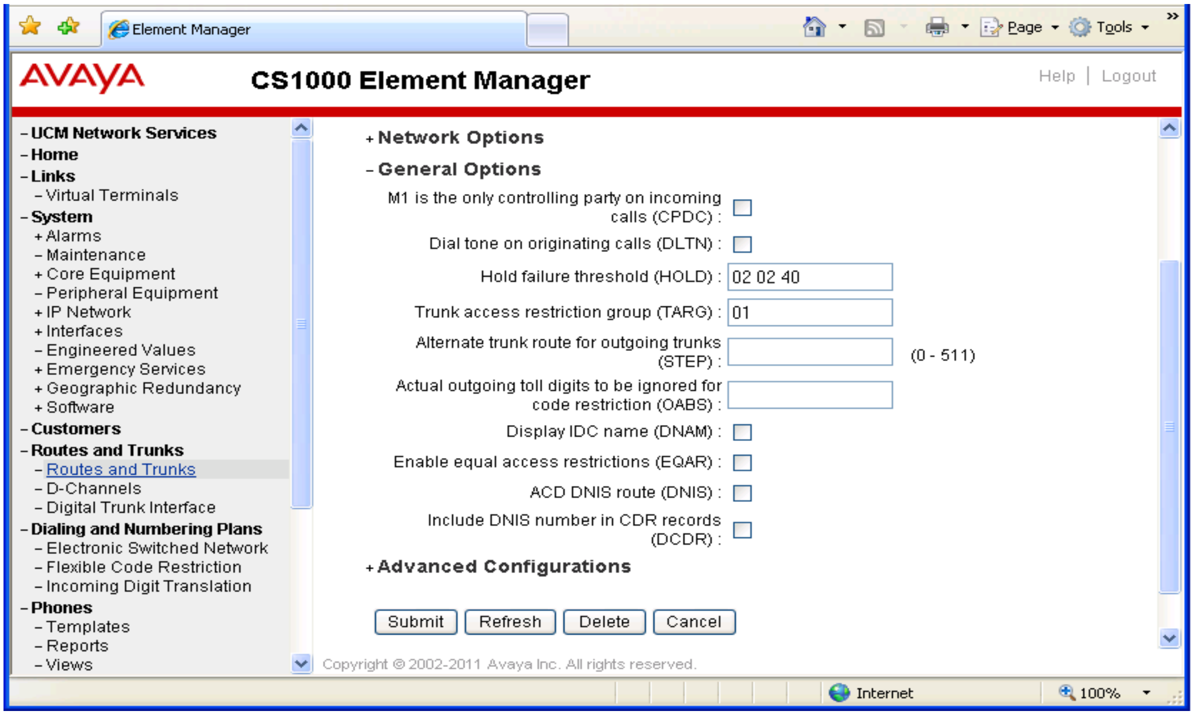

Figure 37: General Options of new Route

Continue expanding the **Advanced Configurations** section and keep its default values as shown in **Figures 38**, **39** and **40**.

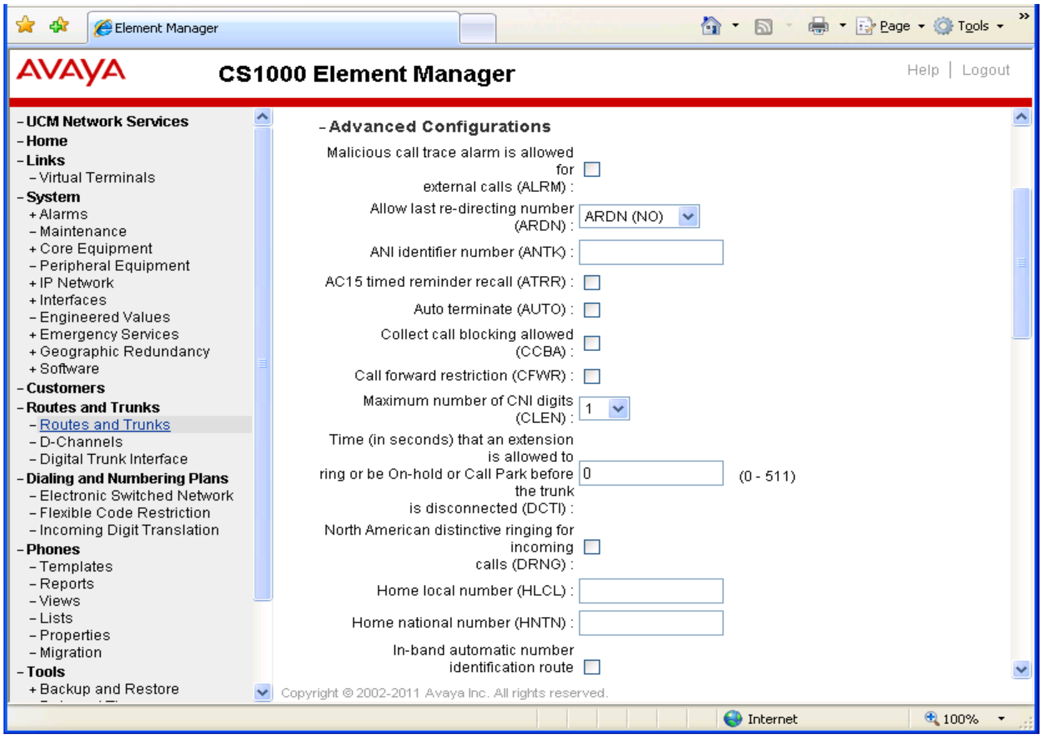

Figure 38: Advanced Configurations of Route

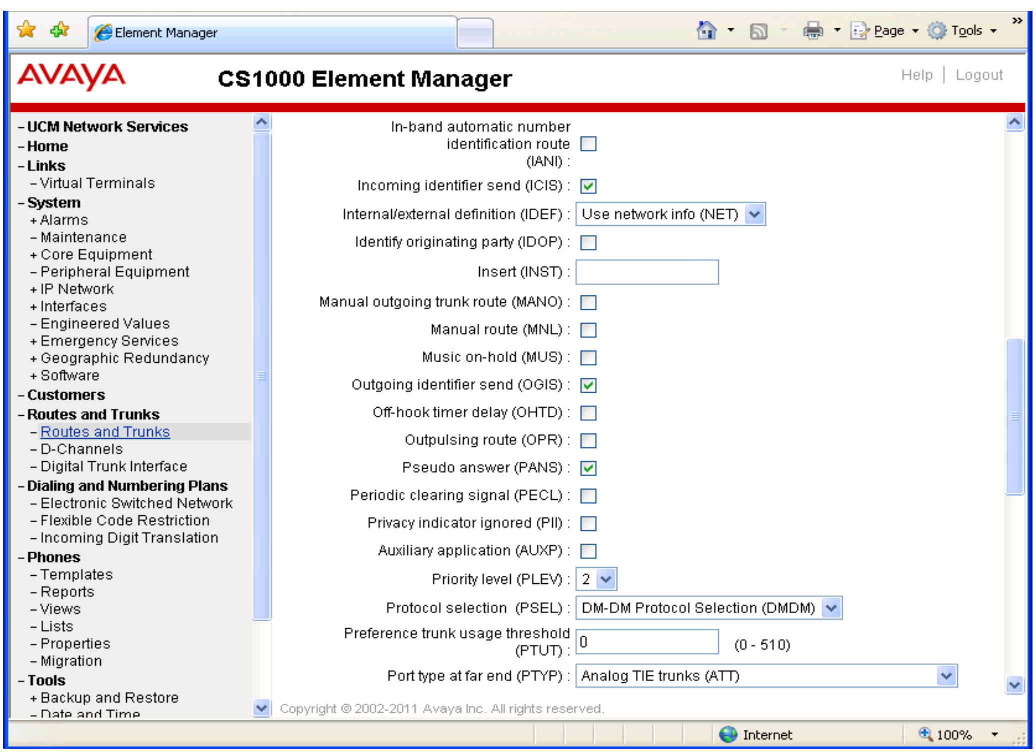

Figure 39: Advanced Configurations of new Route (cont)

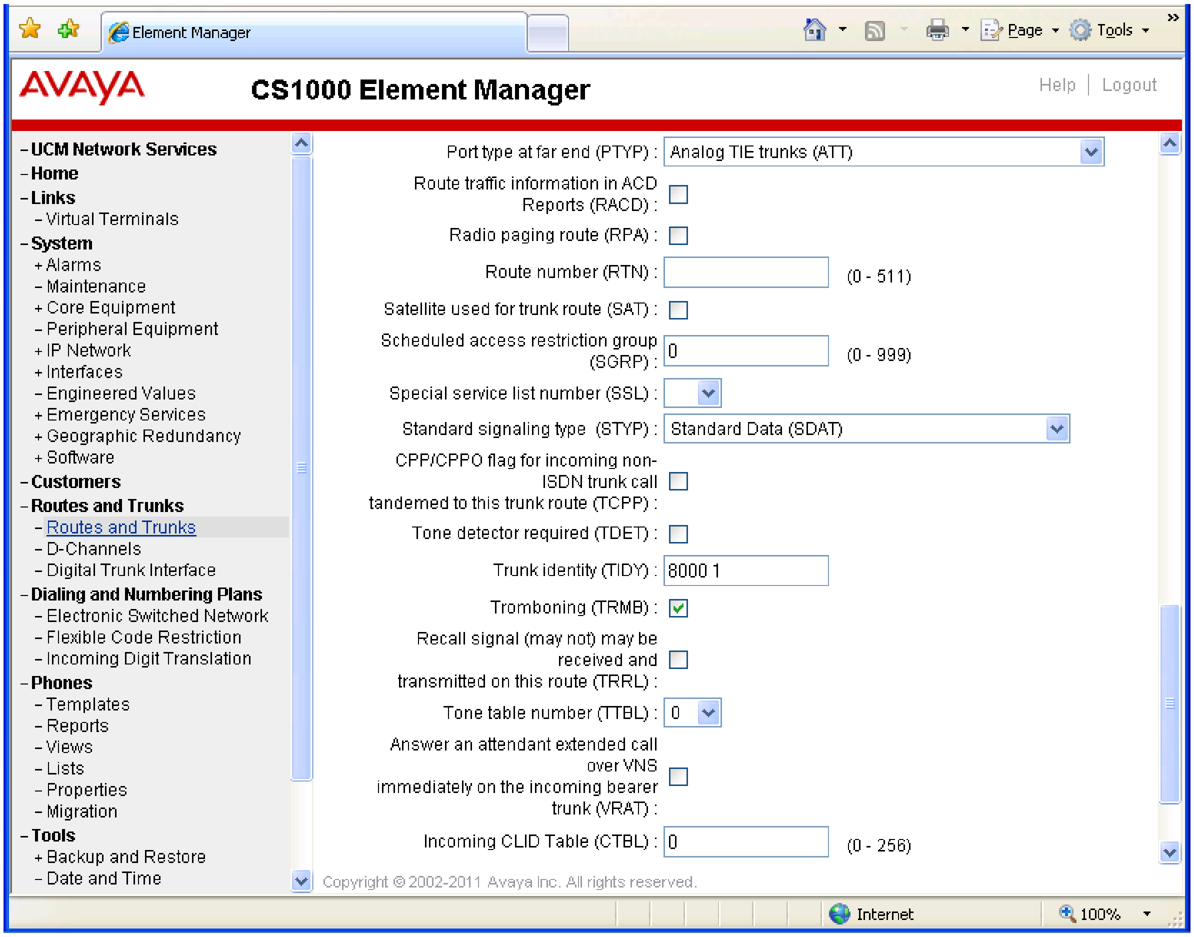

Figure 40: Advanced Configurations of new Route (cont)

6.5. Configure SIP Trunks

To configure a SIP Trunk in the Element Manager, from the homepage of Element Manager navigate to menu **Routes and Trunks > Routes and Trunks**, the **Routes and Trunks** page appears in the right-hand side, under the Customer number (Customer 0) expand the new SIP Route (Route 1 that has been created in the Section 6.4) and click on **Add trunk** button (not shown), the new Trunk page appears as shown in **Figure 41**.

In the **Basic Configuration** section, enter values as shown in **Figure 41** below. Virtual trunks can be created as single or multiple by entering a number in the **Multiple trunk input number** field, normally it an increment of 32, the **Member number** and **Channel ID for this trunk** fields are set to *1* if this is a first virtual trunk of this route, this number is automatically incremented corresponding to the number of trunks created. Click on the **Edit** button of **Class of Service** field to enable necessary class of services of new trunks as shown in **Figures 42** and **43** below. Click on the **Return Class of Service** button as completing enable class of service for new trunks.

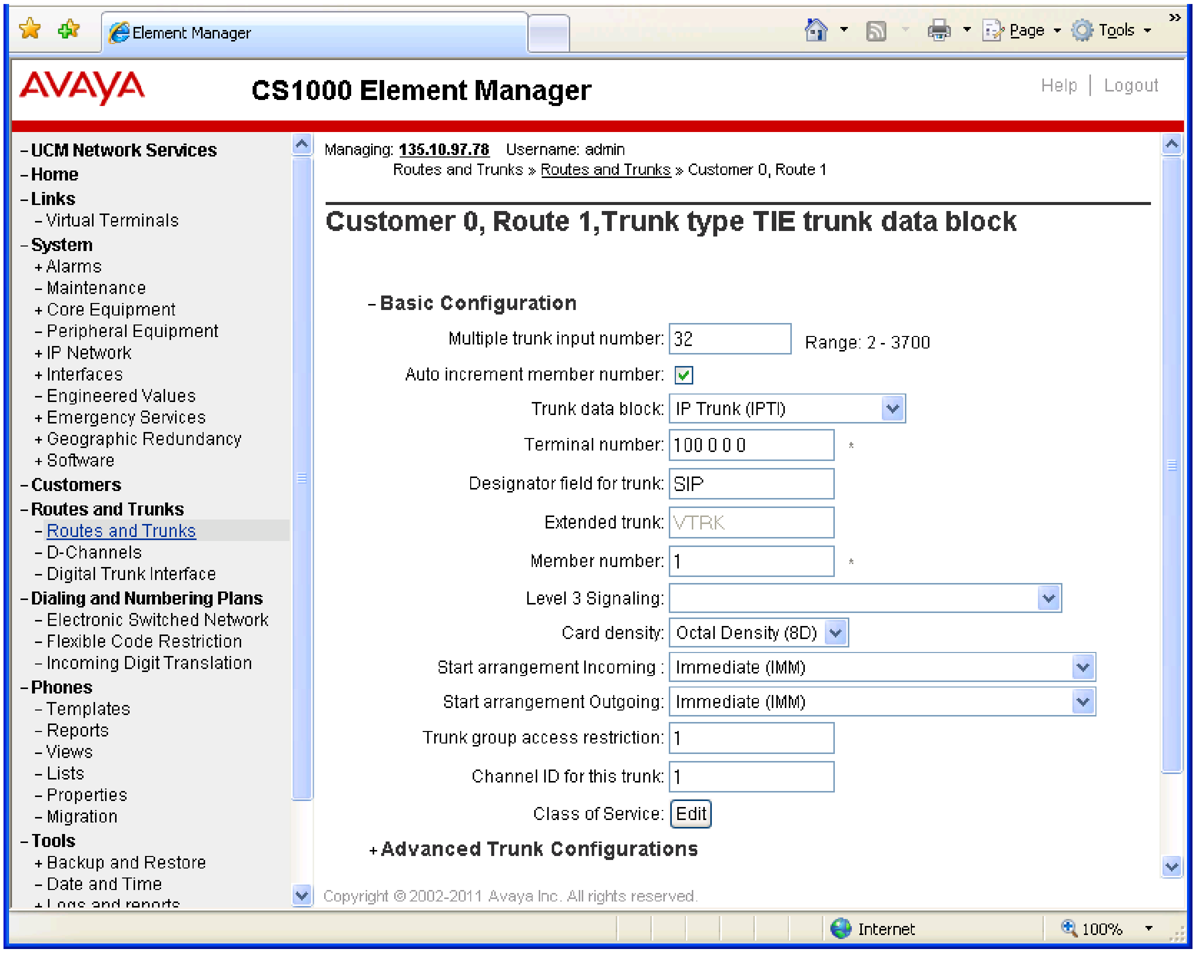

Figure 41: Basic Configuration of new Trunk

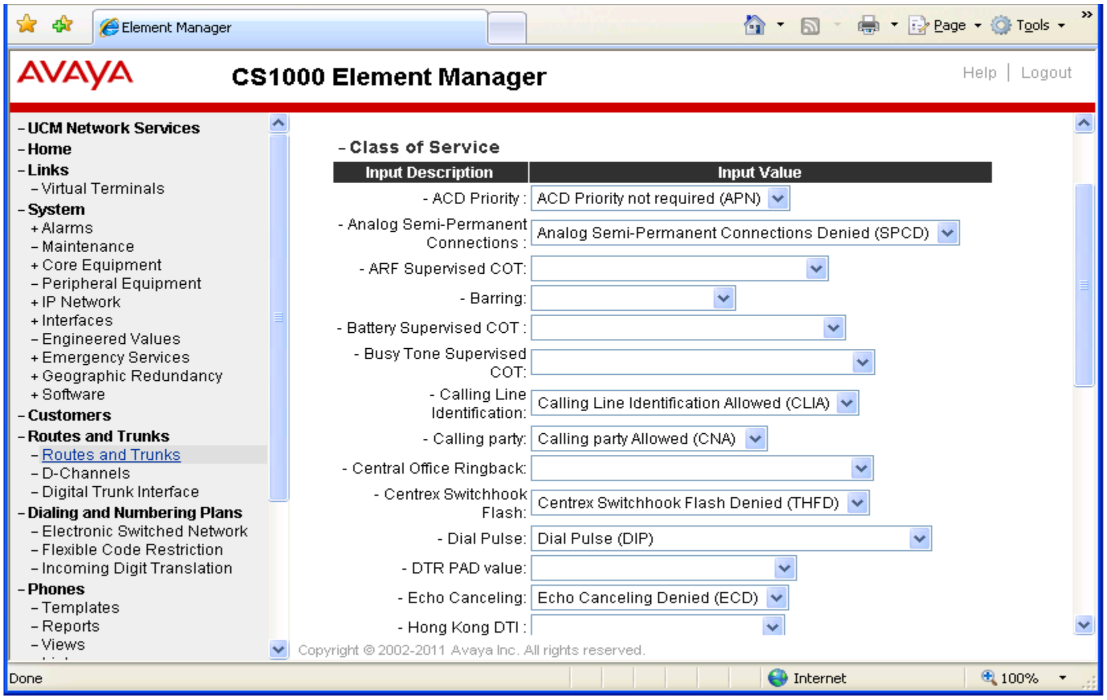

Figure 42: Class of Service of new Trunk

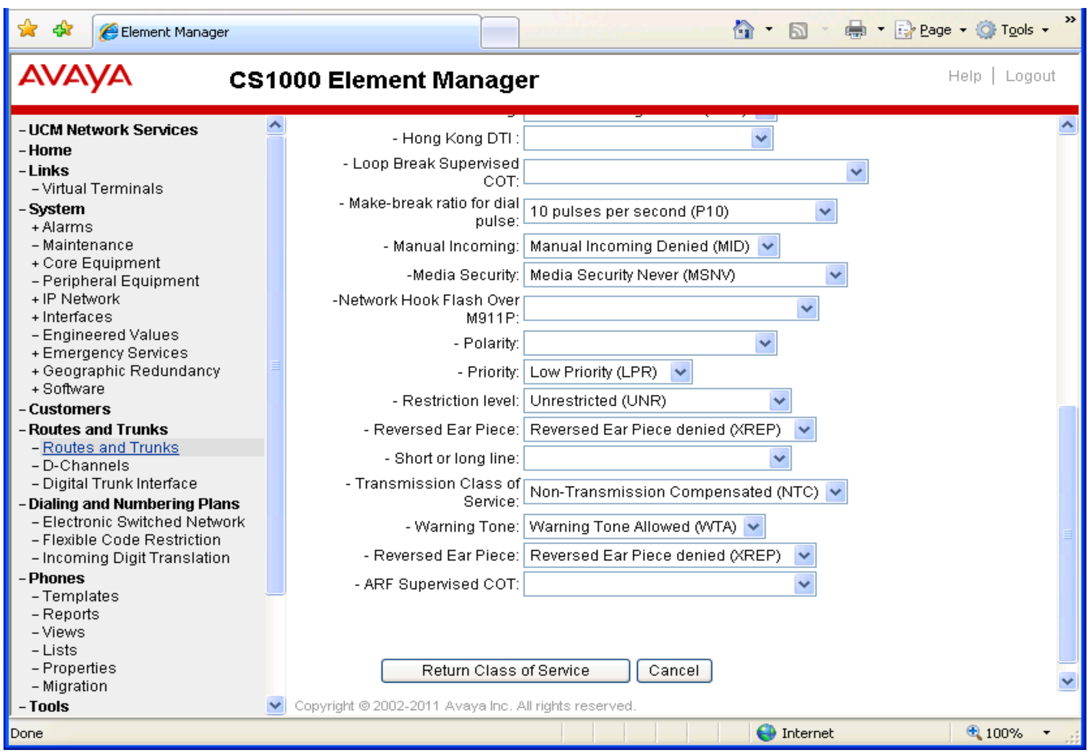

Figure 43: Class of Service of new Trunk (cont)

Continue expanding the **Advanced Trunk Configurations** section and keep its values as shown in **Figure 44** below. Click on **Save** button (not shown) to complete adding new trunks.

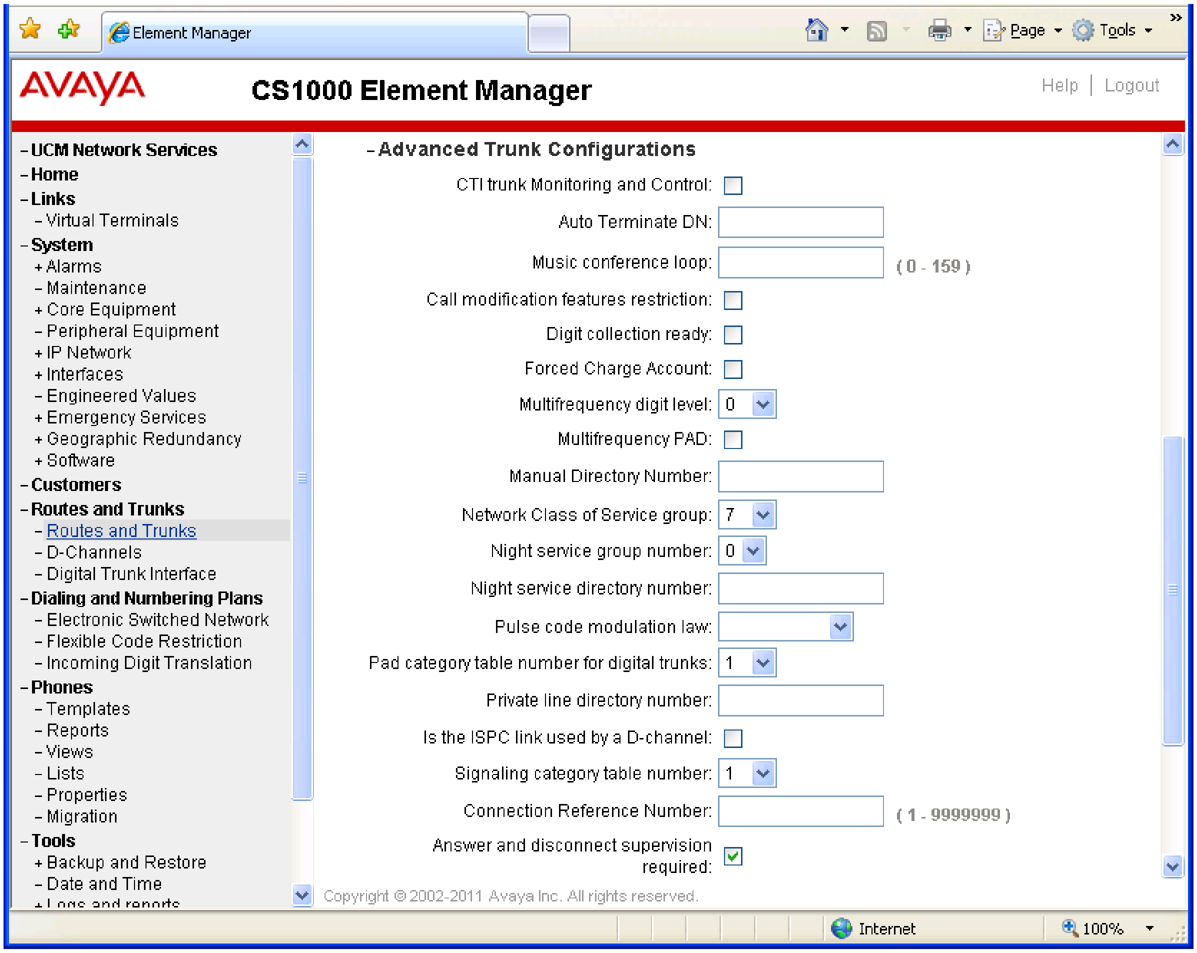

Figure 44: Advanced Trunk Configurations of new Trunk

6.6. Configure CDP Dialing Plan

This section provides the steps on how to create a new Route List Index (RLI) and a new Distant Steering Code (DSC) for the CDP dialing plan.

6.6.1. Configure Route List Index (RLI):

To configure Route List Index, from the home page of Element Manger, navigate to **Dialing and Numbering Plans > Electronic Switched Network**, the **Electronic Switched Network (ESN)** page appears as shown in **Figure 45** below.

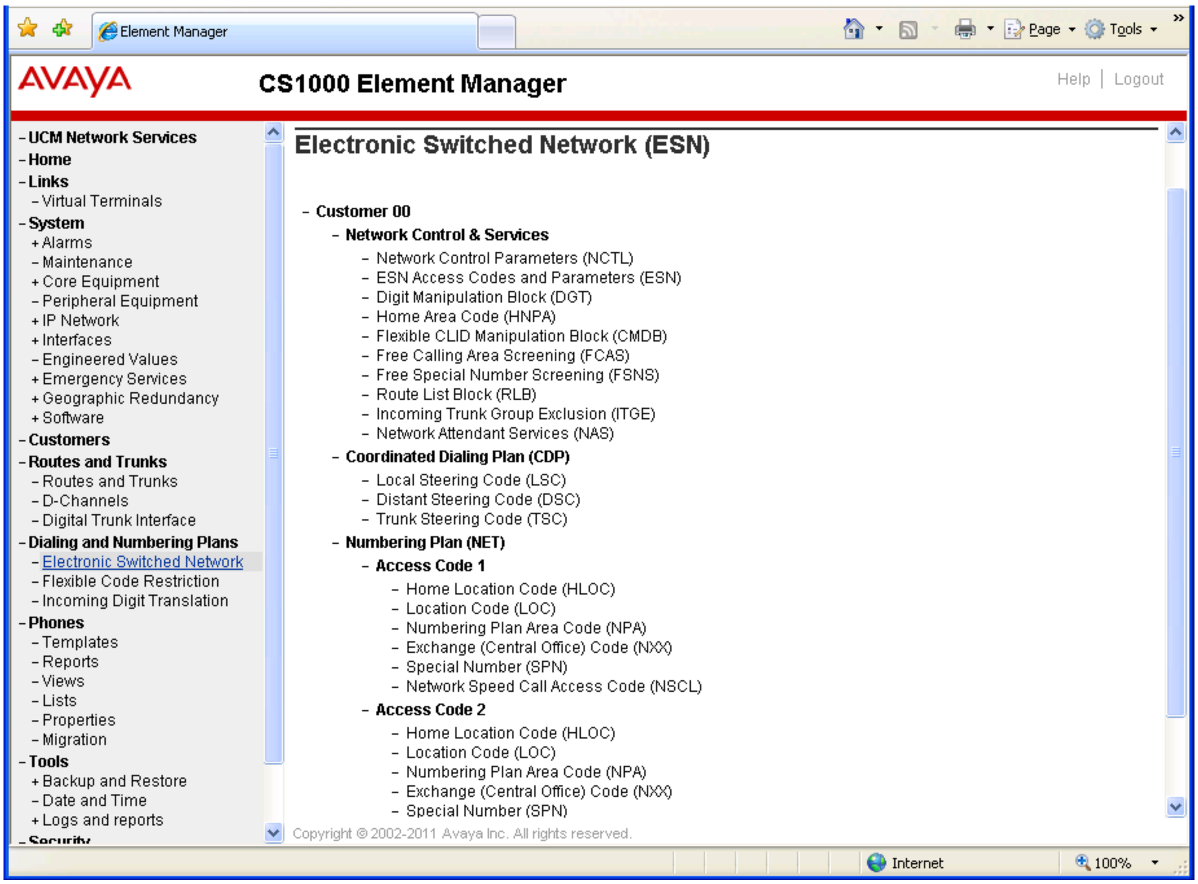

Figure 45: Electronic Switched Network (ESN) page

Click on the **Route List Block** (**RLB**) link of the **Electronic Switched Network (ESN)** page, the **Route List Blocks** page appears as shown in **Figure 46**. To create a new entry for route list index, enter a number, e.g. *10*, in the **Please enter a route list index** box and then click on **to Add** button.

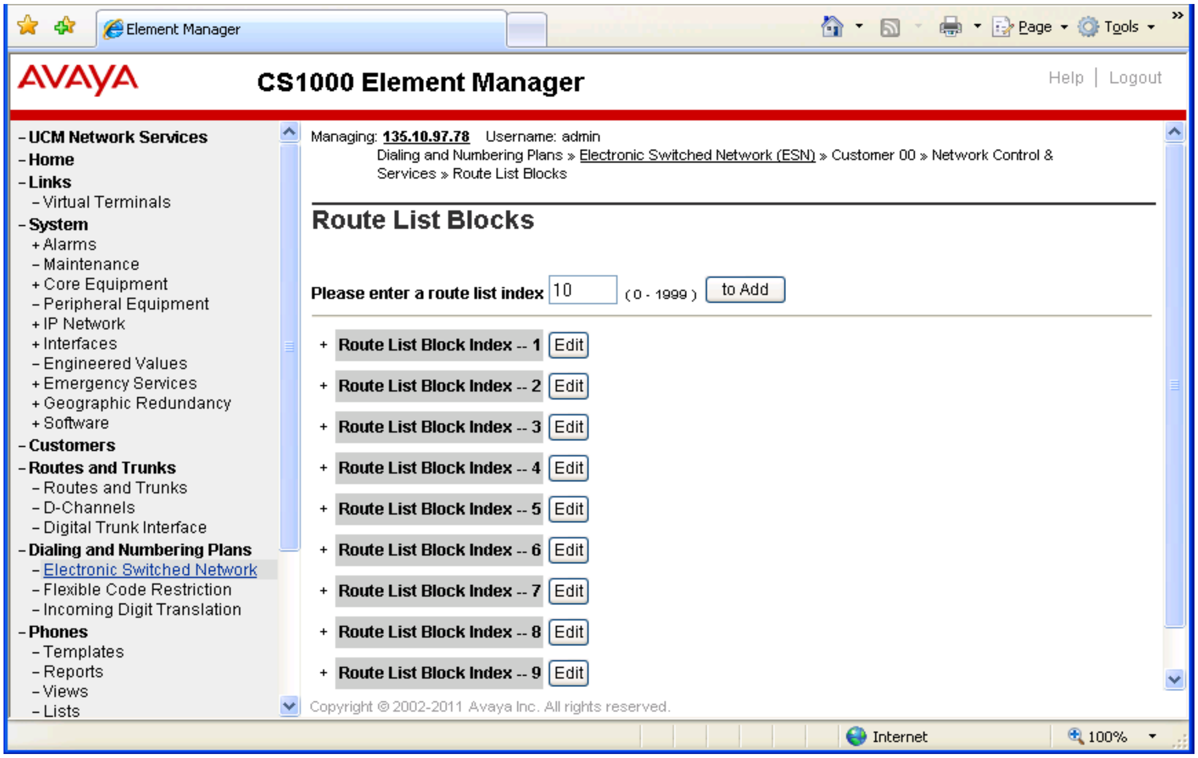

Figure 46: Route List Blocks page

The **General Properties** and **Indexes** sections of new route list index appear as shown in **Figure 47** below. Keep all values as default.

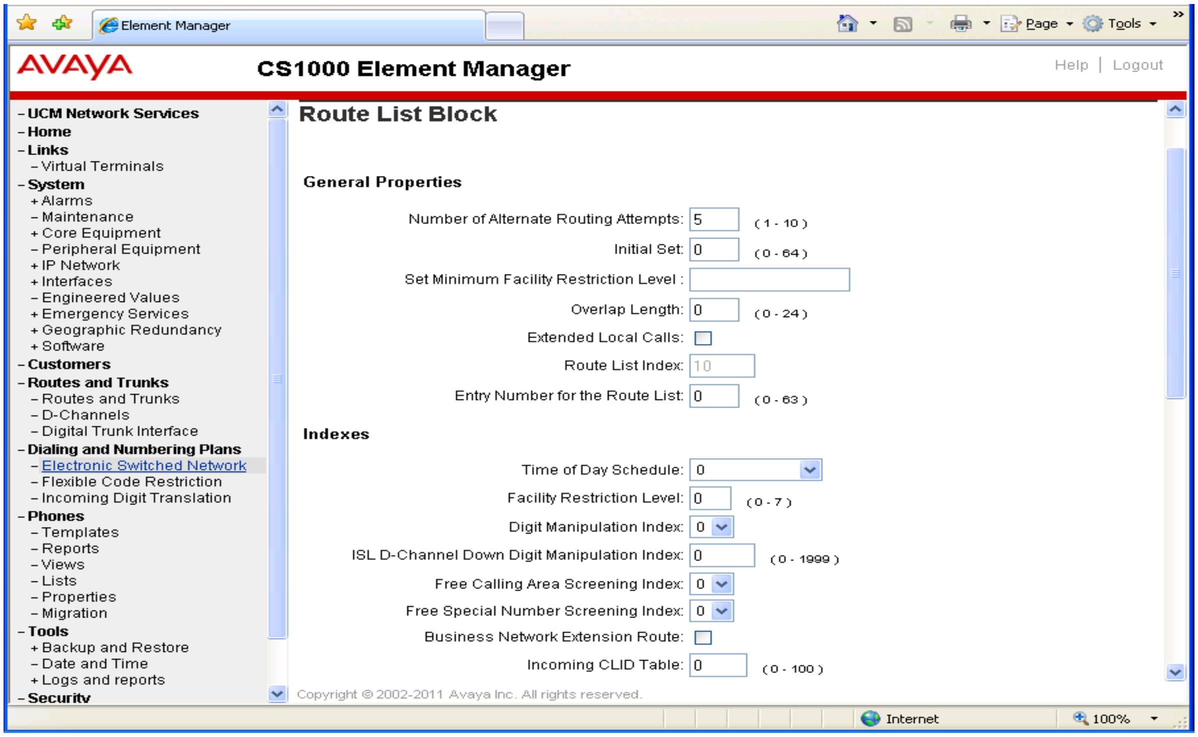

Figure 47: General Properties and Indexes of Route List Block page

In the **Options** and **VNC Options** sections that are shown in **Figure 48**, keep all values at default but select the **Route Number** in the dropdown list corresponding with the SIP Route that has been created in **Section 6.4**. Click **Submit** button to complete adding new route list index.

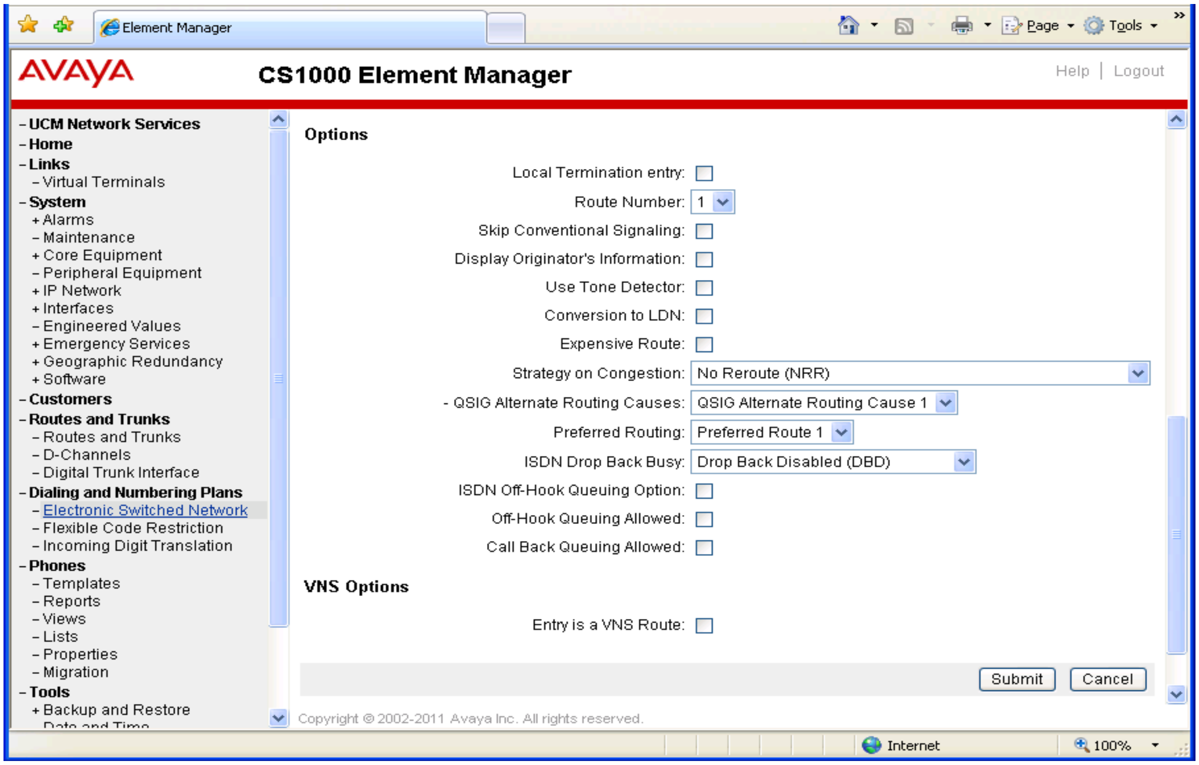

Figure 48: Options and VNS Options sections of Route List Blocks page (cont)

6.6.2. Create a Distant Steering Code (DSC)

To create a new distant steering code, from the home page of Element Manager navigate to **Dialing and Numbering Plans > Electronic Switched Network > Coordinated Dialing Plan (CDP) > Distant Steering Code (DSC)**, the **Distant Steering Code List** page appears as shown in **Figure 49** below, select **Add** in the dropdown menu and then enter the DSC code *73100* in the **Please enter a distant steering code** field and then click on to **toAdd** button.

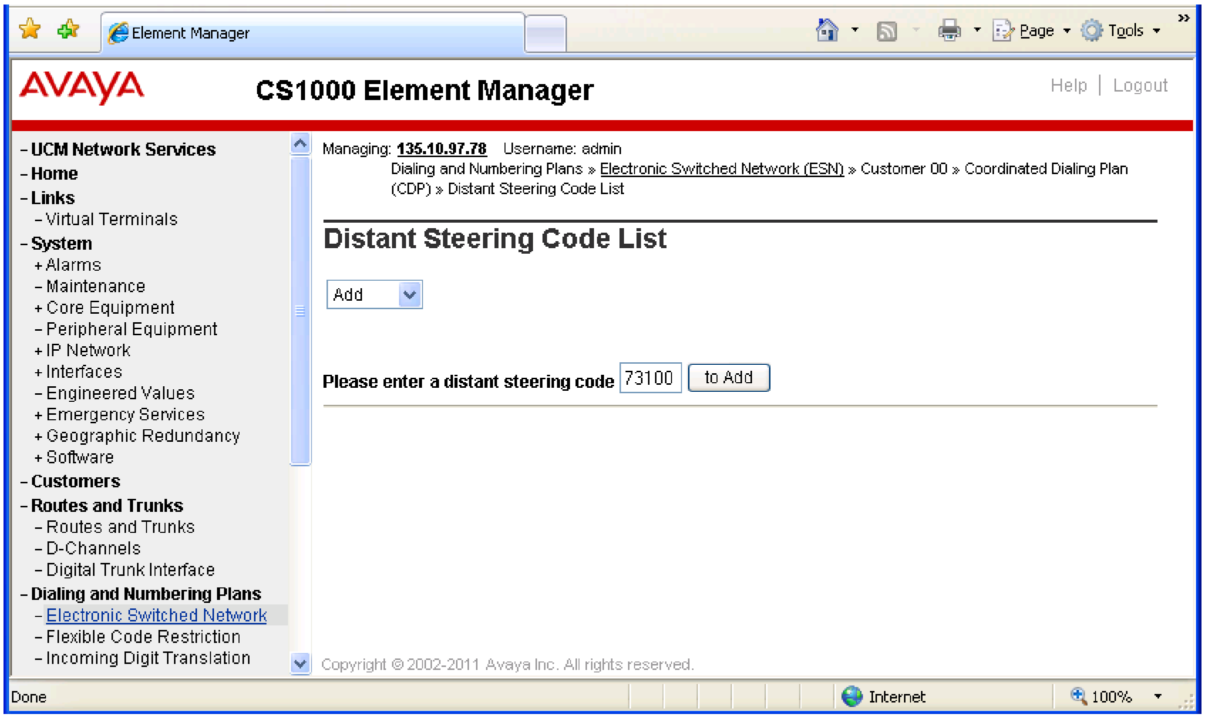

Figure 49: Distant Steering Code List page

The **Distant Steering Code** page appears as shown in **Figure 50**, enter *5* in the field **Flexible Length number of digits**, because the length of dialled number to Polycom RMX 4000 system is 5 digits, if 4 or 3 digits is planned, enter the corresponding length of digit in this field and select the route list index *10* that has been created above in the **Route List to be accessed for trunk steering code** (RLI 10) dropdown list. Click on the **Submit** button to complete adding new distant steering code.

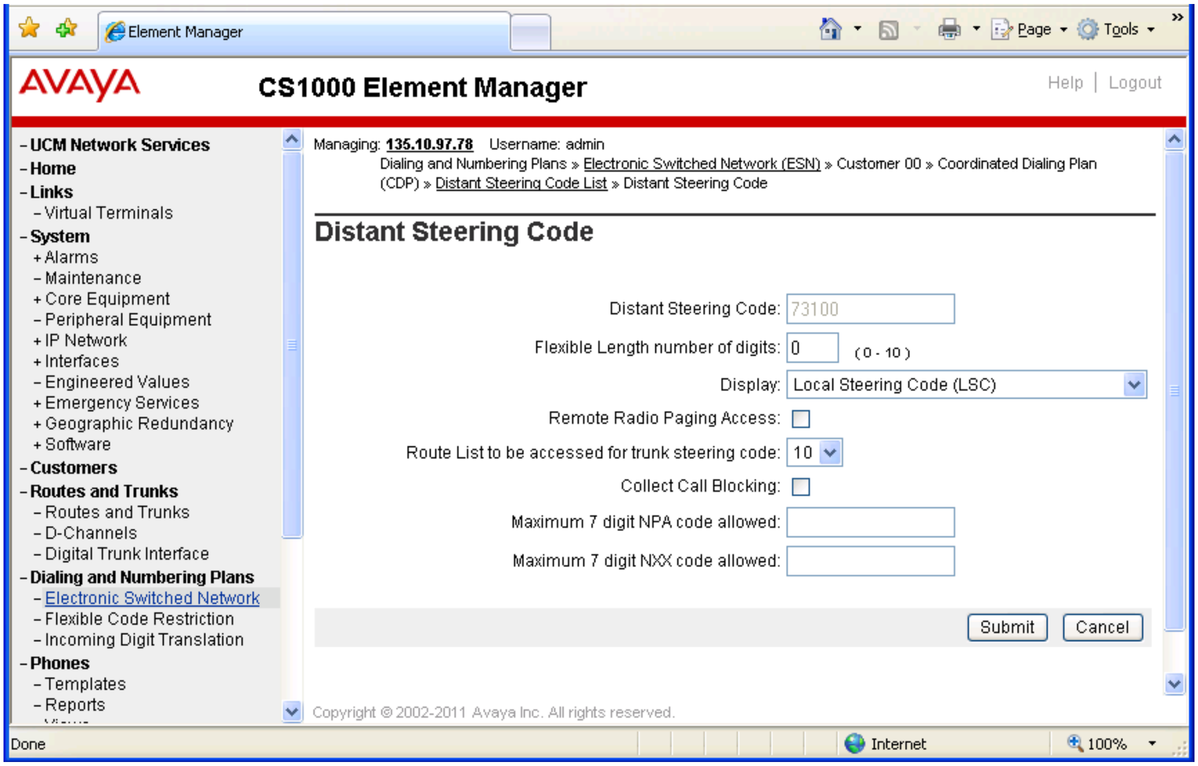

Figure 50: Distant Steering Code page

7. Configure Avaya CallPilot® Through-Dial

This document assumes that the Avaya CallPilot messaging system was properly installed, configured and administered as per the product document, for more information about how to install, configure and administer the CallPilot system please refer to **Section 11 [8].** This section provides the steps on how to configure the through dial feature by using the Application Builder application.

The Application Builder software needs to be installed on a desktop PC. From the desktop PC, launch the CallPilot Manager webpage by entering the URL *http://CallPilotipaddress/cpmgr* in the address bar of an internet browser, and then enter the mailbox number *000000* and its password (not shown). Click on the **Login** button to log in, **Figure 51** below shows CallPilot Manager webpage.

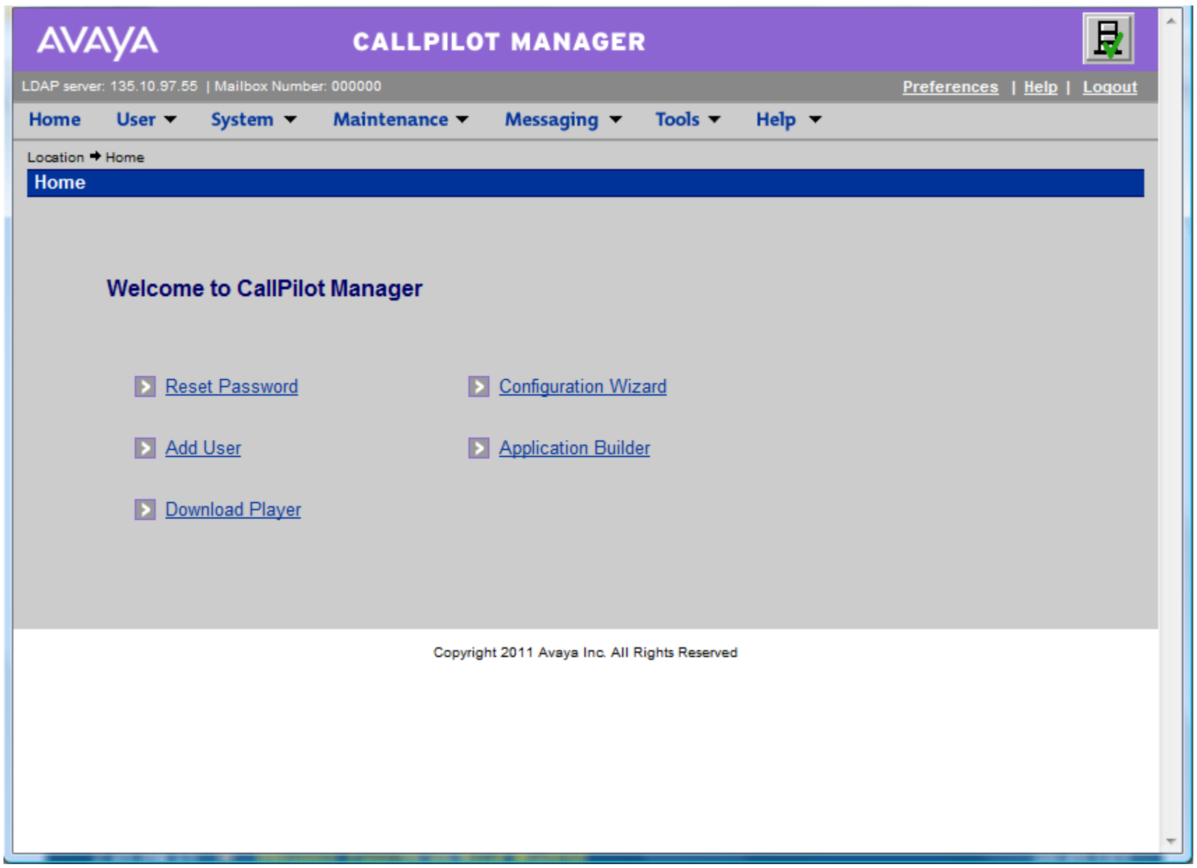

Figure 51: CallPilot Manager Home page

Click on the **Application Builder** link, please note the Application Builder link only displayed on the PC that has the Application Builder software installed, the Login window of Application Builder appears as shown in **Figure 52**, enter the User ID *000000* and password as the same as used to login to the CallPilot Manager webpage. Click **OK** button to login.

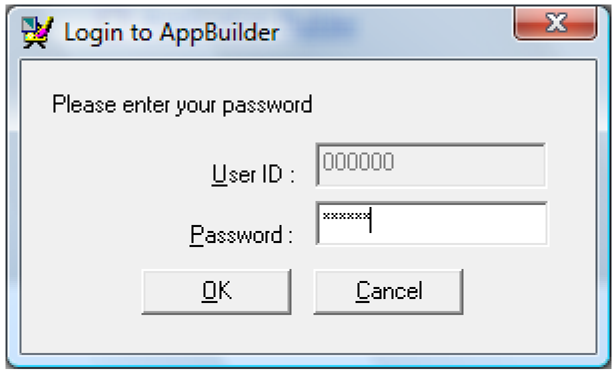

Figure 52: Login window of Application Builder

The **Application Builder** window opens as shown in **Figure 53**.

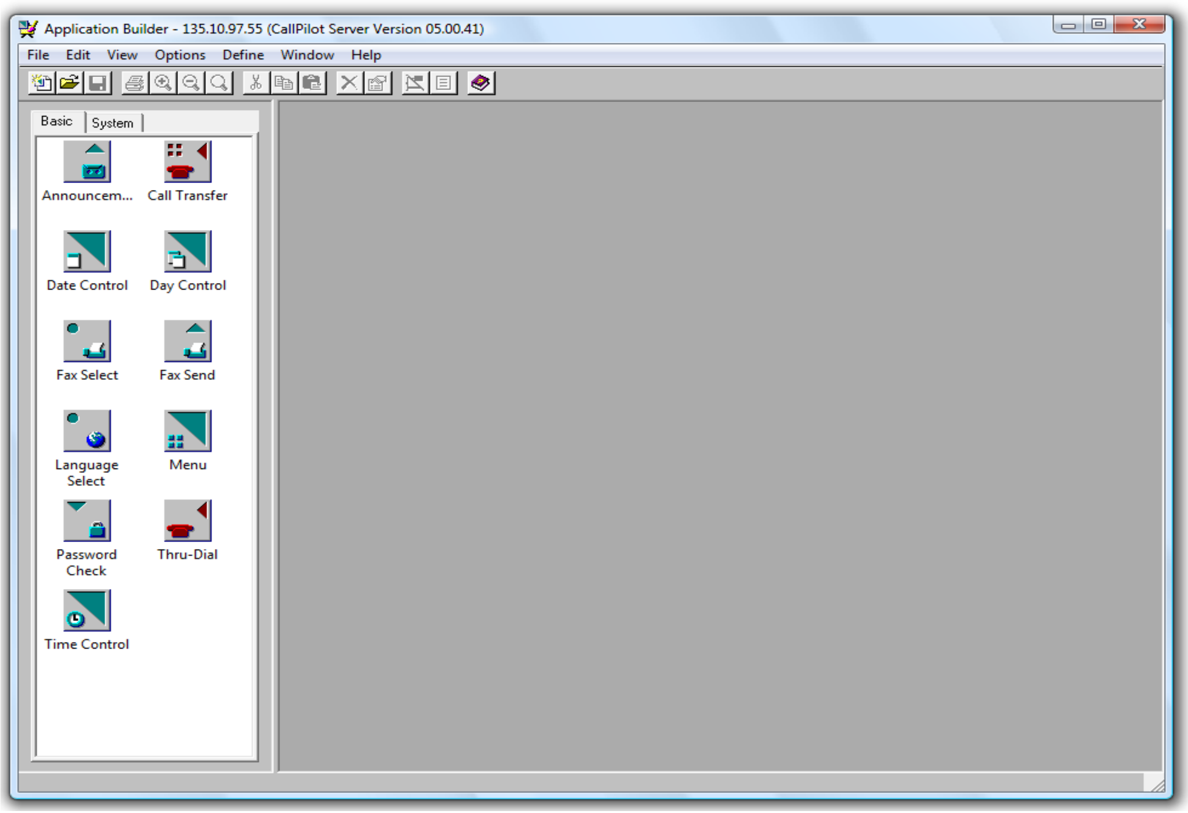

Figure 53: Application Builder window

From the **Application Builder** window, navigate to menu **File** > **New** to create a new application; the **New** application window appears as shown in **Figure 54**. Enter a name in the **File name** box, for example *Thru_2_RMX*, and then click on the **New** button.

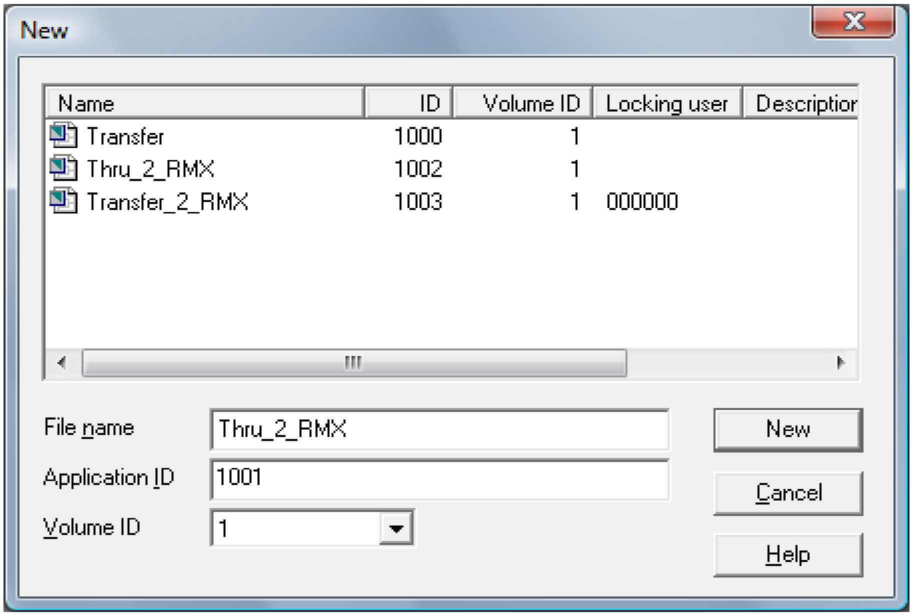

Figure 54: New window of Application Builder

The Thru_2_RMX application window is created and opened as shown in **Figure 55** below.

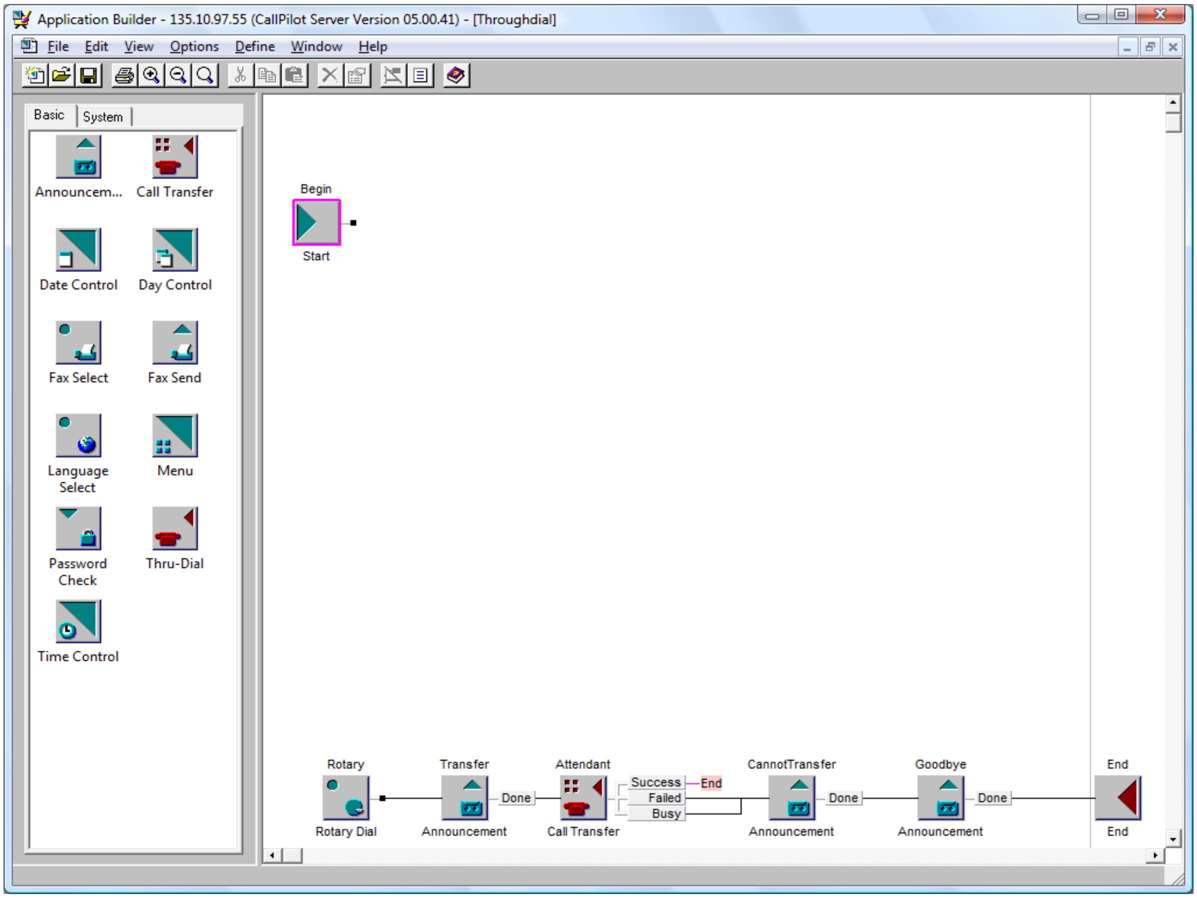

Figure 55: New application created and ready to build

Drag the **Thru-Dial** feature icon from the **Basic** tab in left-hand side into the working area of **Thru_2_RMX** application and customize it to become a finished application as shown in **Figure 56** below.

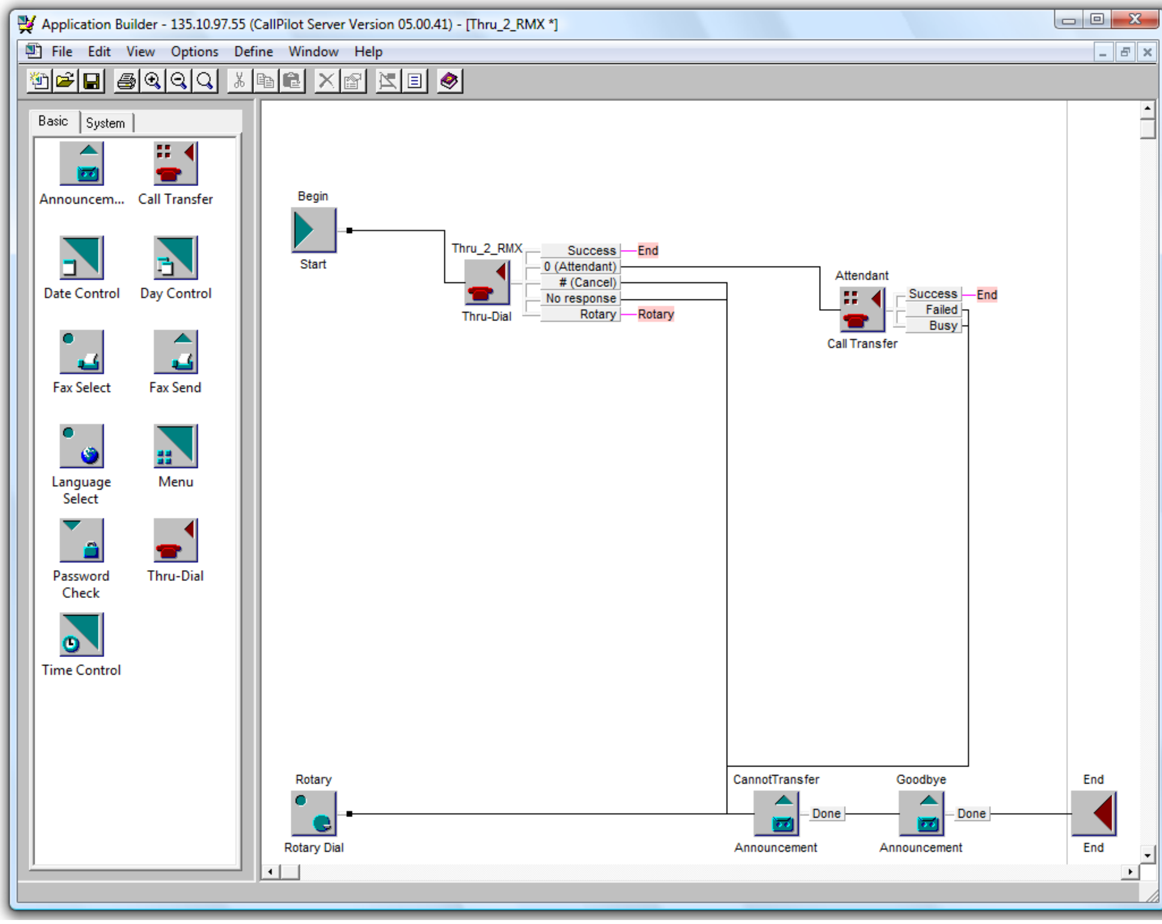

Figure 56: The finished application of Thru-dial feature

Save the **Thru_2_RMX** finished application, close the Application Builder and go back to the CallPilot Manager webpage and navigate to menu **System > Service Directory Number**, the **Service Directory Number** page appears as shown in **Figure 57**.

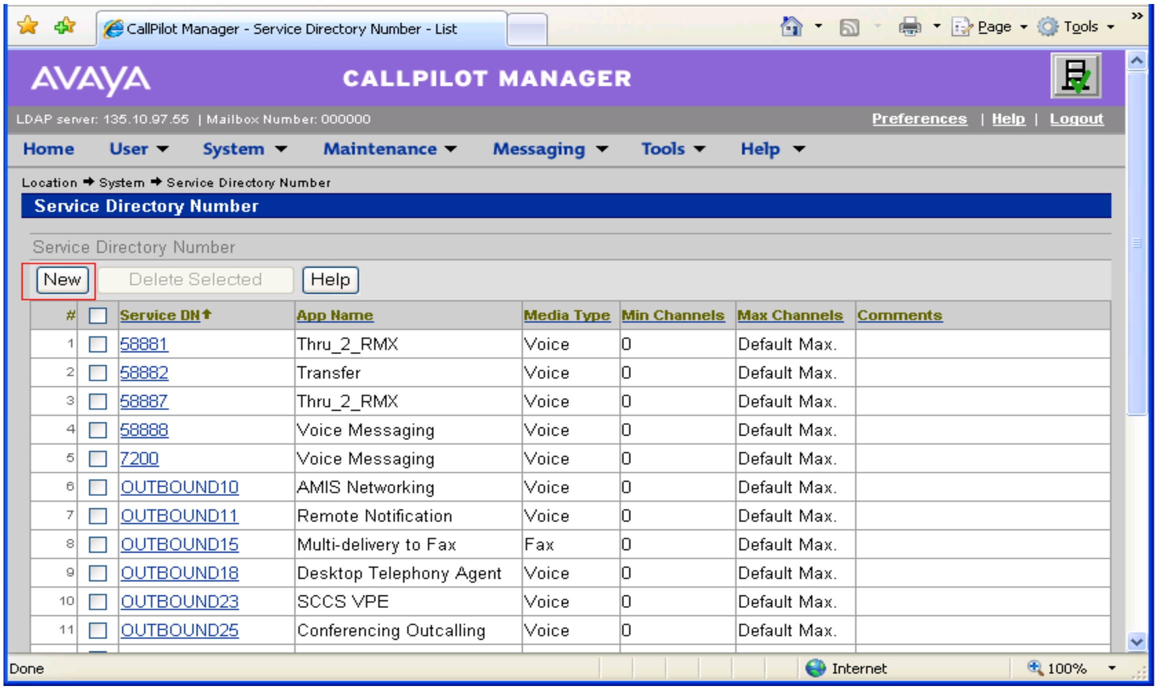

Figure 57: CallPilot Service Directory Number page

Click on the **New** button to add a new service directory number (SDN), the new SDN Details page appears as shown in **Figure 58** below, enter the SDN *58881* (This DN is defined in overlay (LD) 23 of CS1000 Call Server) in the **Service DN** box and select *Thru_2_RMX* application that was created above in the **Application Name** field. Click on **Save** button to complete adding the new Service DN.

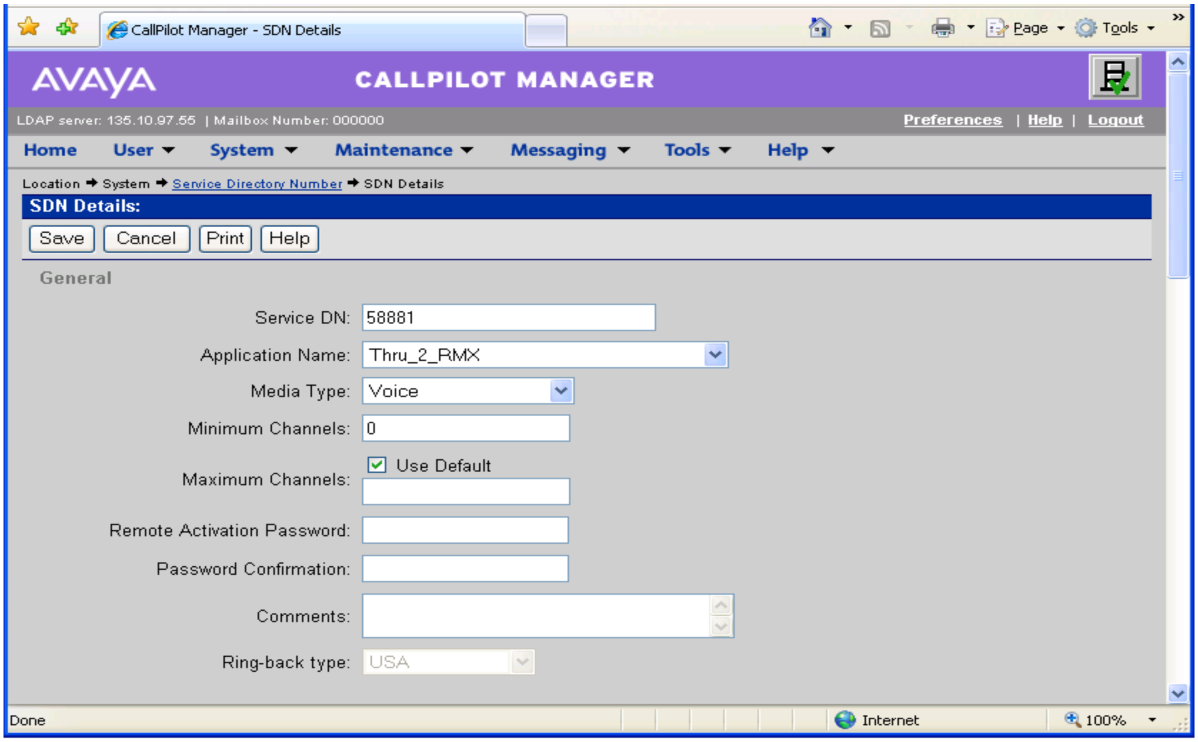

Figure 58: Adding new SDN

8. Configure Polycom RMX 4000

This document assumes that the Polycom RMX 4000 system was properly installed and configured by a Polycom Engineer. This section provides the steps to configure Polycom RMX 4000 working with Avaya Communication Server 1000 system. For more information on how to configure and administer the Polycom RXM 4000 please refer to **Section 11[8]**.

To log in to the Polycom RMX 4000 Web Client, launch a web browser from a desktop PC which is able to reach to RMX 4000 server and type the management IP address of the RMX 4000 server into the address bar. The Login page of Polycom RMX 4000 appears as shown in **Figure 59**.

Use the username *SUPPORT* and its password to log in the RMX 4000 Web Client. Click on the **Login** button to log in to the RMX 4000.

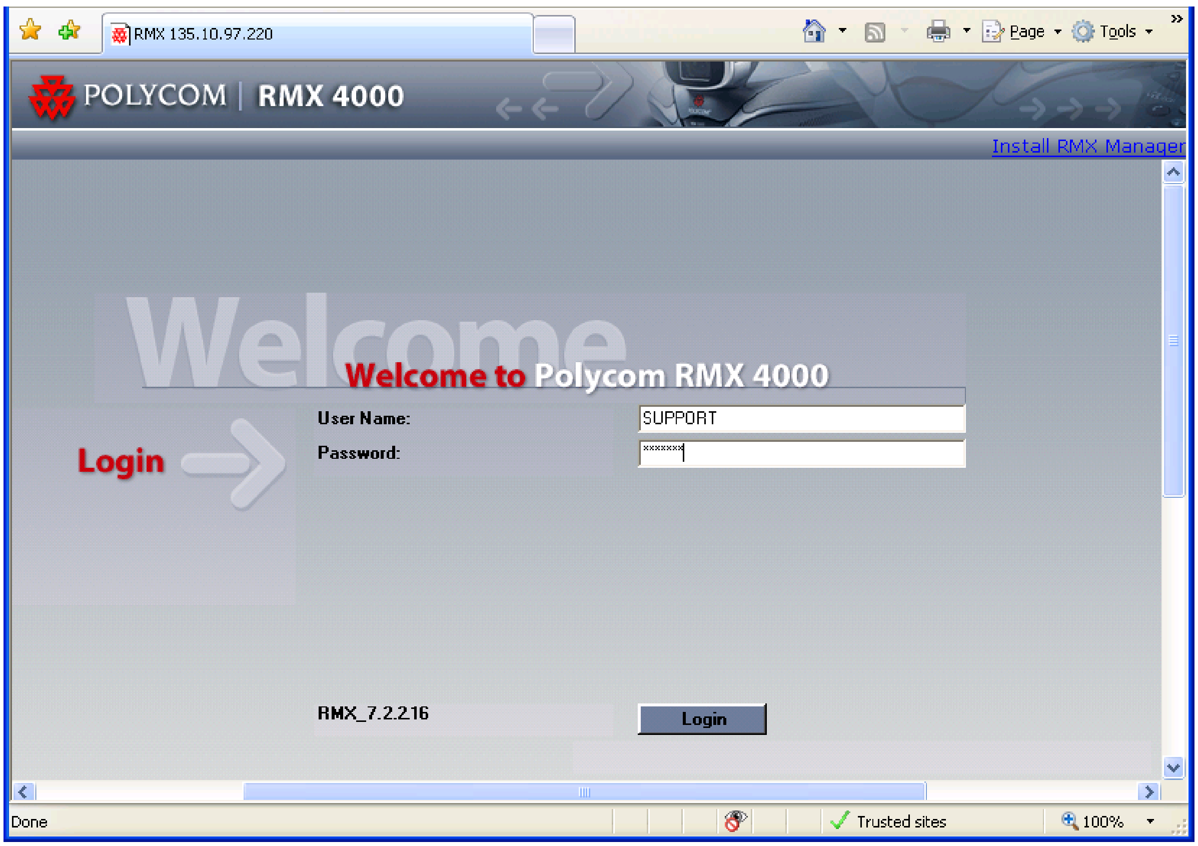

Figure 59: Polycom RMX 4000 Web Client Login Page

Figure 60 below shows the home page of RMX 4000 Web Client after logged in.

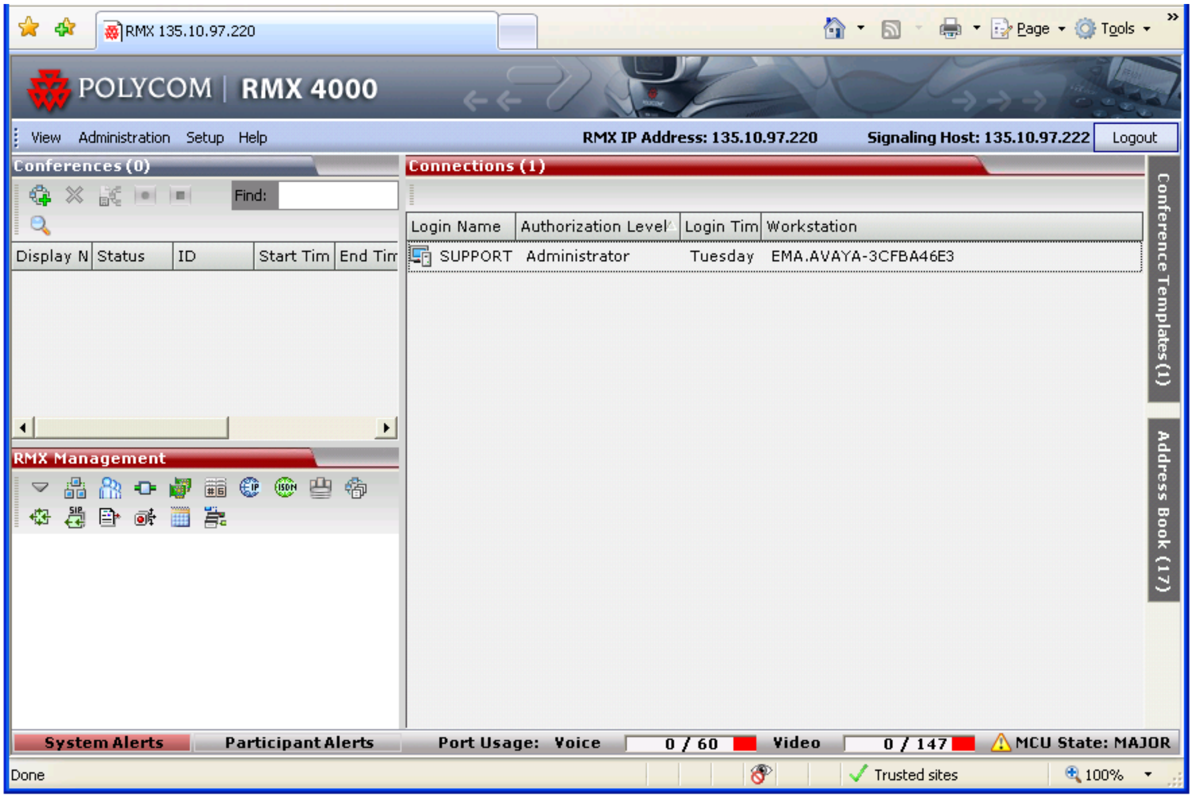

Figure 60: Polycom RMX 4000 Web Client Home Page

8.1. Configure IP Network Service

To configure **IP Network Service**, in the homepage of RMX 4000 Web Client as shown in **Figure 60**, click on the **IP Network Service** icon under the **RMX Management** window. The **IP Network Services** window appears on the right-hand side as shown in **Figure 61**.

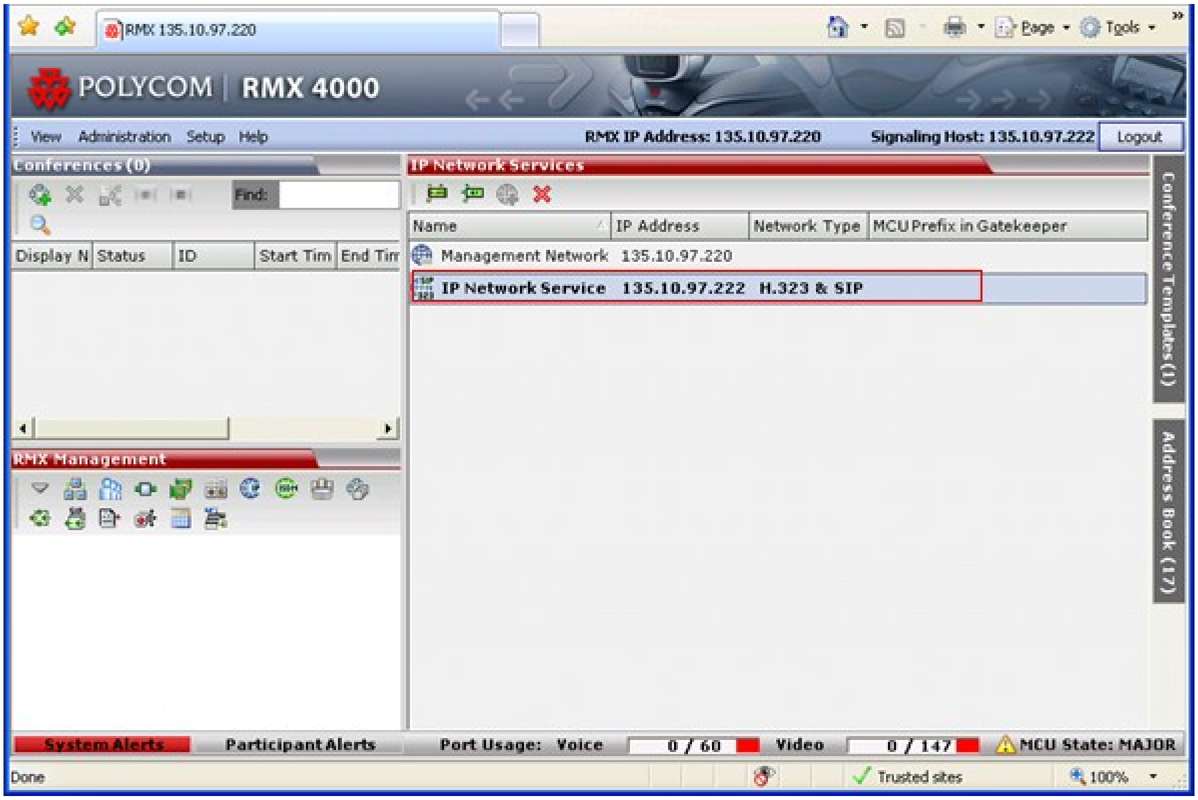

Figure 61: RMX IP Network Service

Double-click on the **IP Network Service** name, the **IP Network Service Properties** window with **IP** section appears as shown in **Figure 62**.

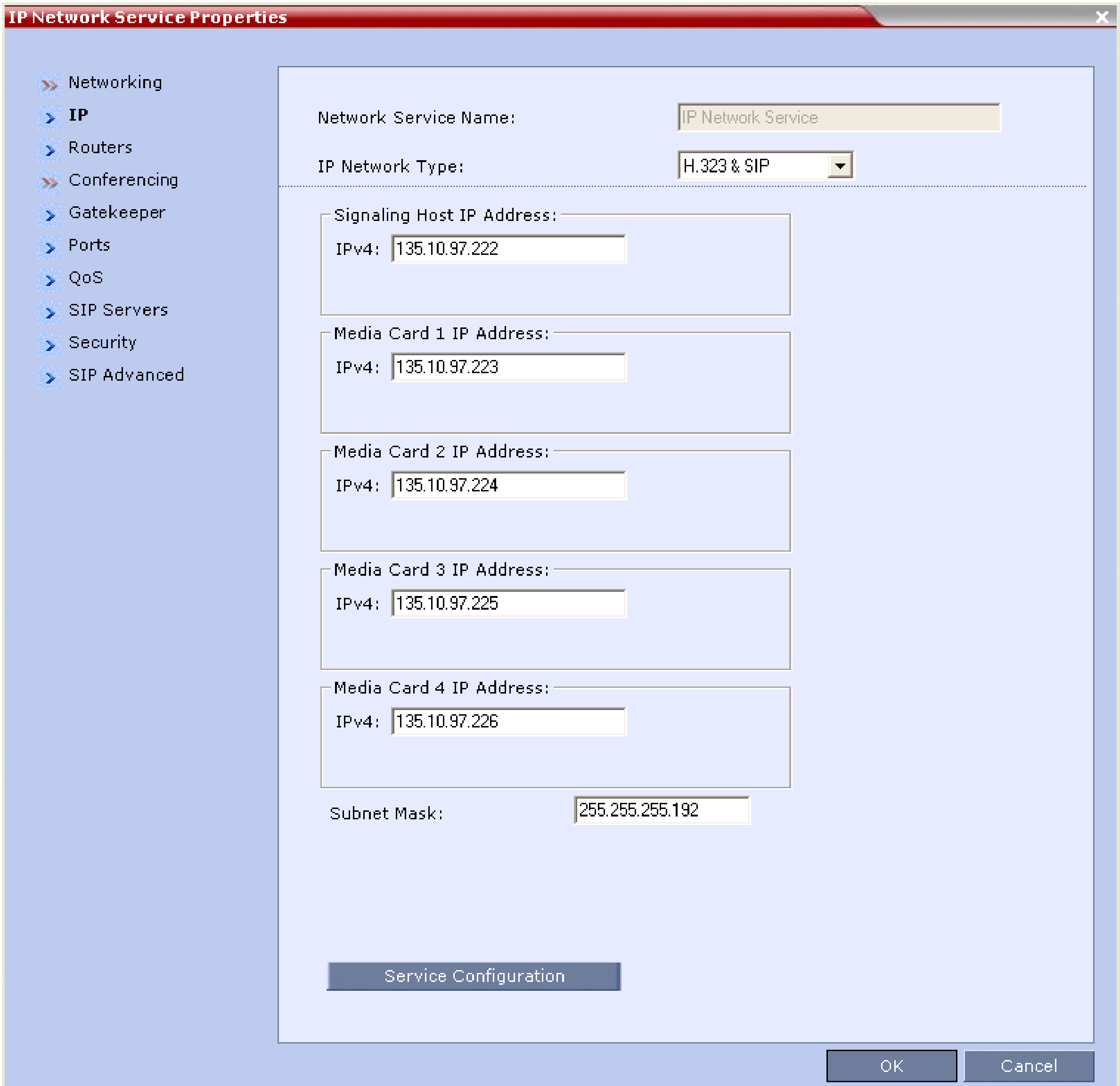

Figure 62: IP Network Service Properties Window

Click on the **SIP Servers** tab on the left-hand side of **IP Network Service Properties** window, the SIP Servers setting section appears in the righ-hand side as shown in **Figure 63**.

In the **SIP Server** field select as *Specify* and the **Tranport Type** select *UDP*.

In the **SIP Servers** table, enter the **Server IP Address or Name** as signaling IP address of Session Manager *135.10.97.198*, **Server Domain Name** as *bvwdev.com* as defined in the **Section 5.1**, and the **Port** *5060*.

In the **Outbound Proxy Servers** table: Enter the **Server IP Address** or **Name** as signaling IP address of Session Manager *135.10.97.198* and the **Port** *5060*. Keep other fields as defaults.

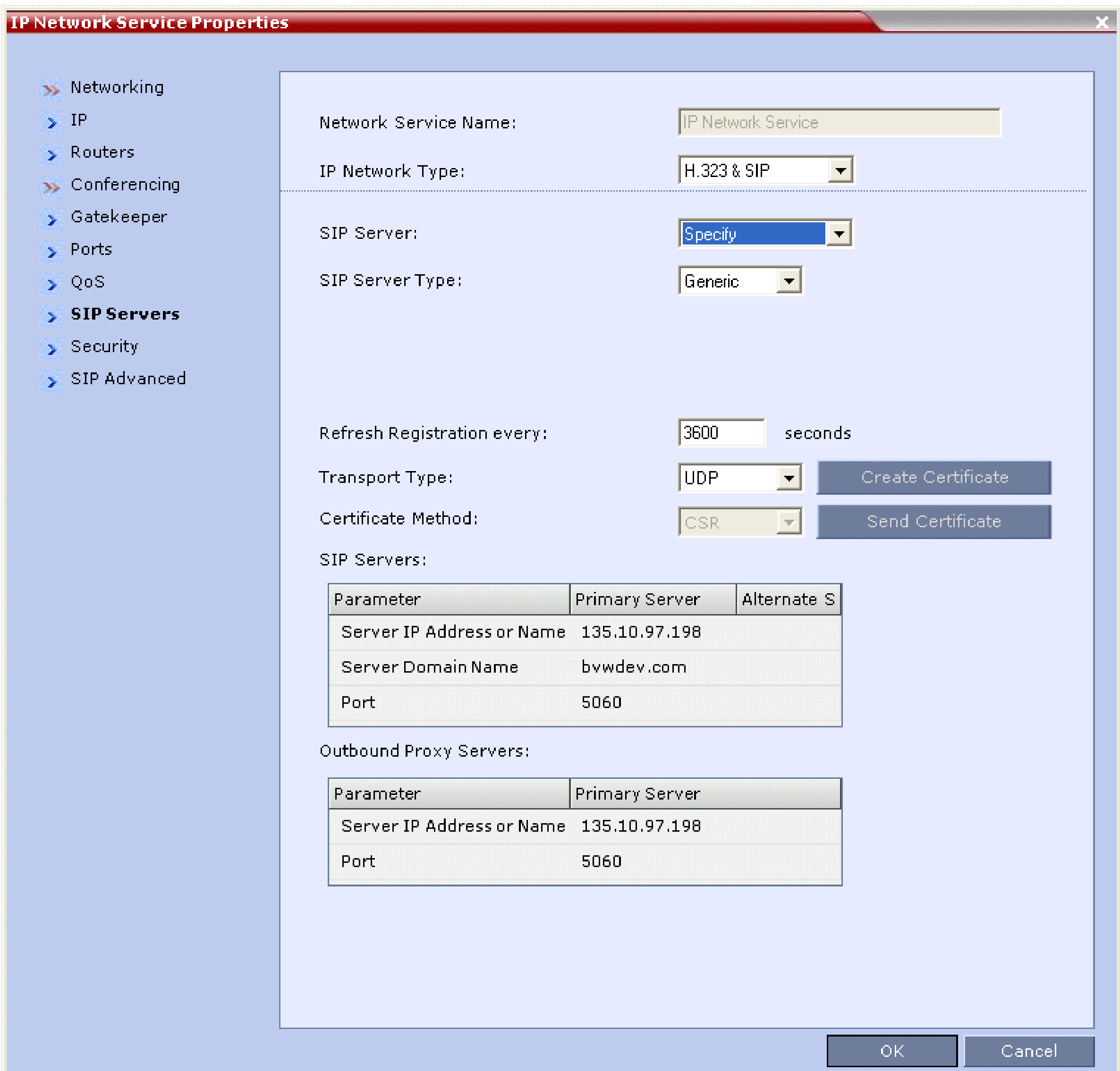

Figure 63: SIP Servers section of IP Network Service

Click on the **Gatekeeper** tab, the **Gatekeeper** content appears in right-hand side of the **IP Network Service Properties** window appears as shown in **Figure 64** below.

Fisrtly, select *Specify* in th **Gatekeeper** field, enter the signaling IP address of Session Manager *135.10.97.198* in the **IP Address or Name** box of **Primary Gatekeeper** field, enter the DN *73100* in the **MCU prefix in Gatekeeper** field and check the **Register as Gateway** checkbox. Keep other fields as default.

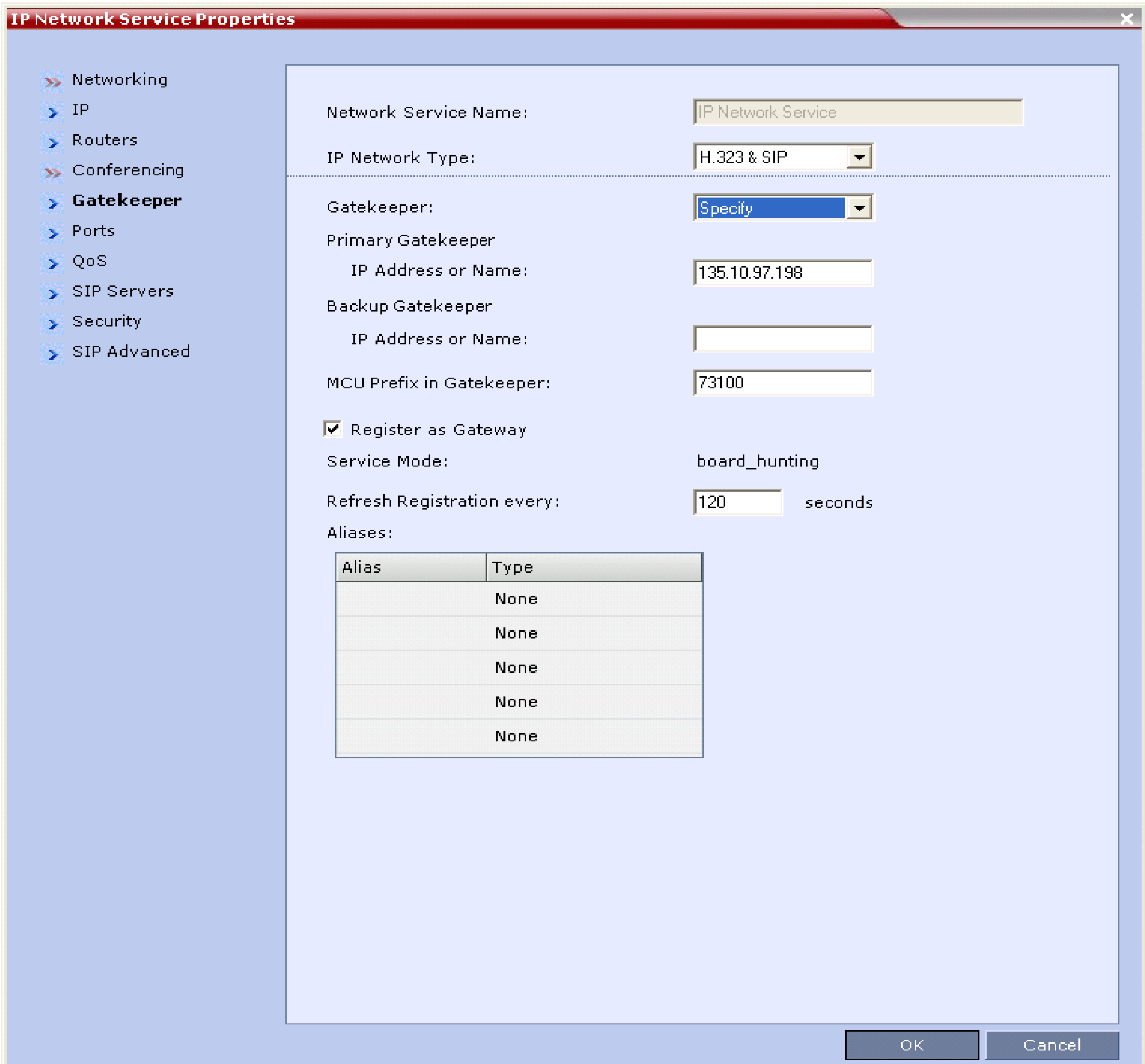

Figure 64: Gatekeeper section of IP Network Service

Secondly, select *Off* in the **Gatekeeper** field as shown in **Figure 65** below.

Note:
The **MCU Prefix in Gatekeeper** field is very important for RMX 4000 to work properly and it cannot be blank, instead main number of RMX 4000 system must be present in this field.

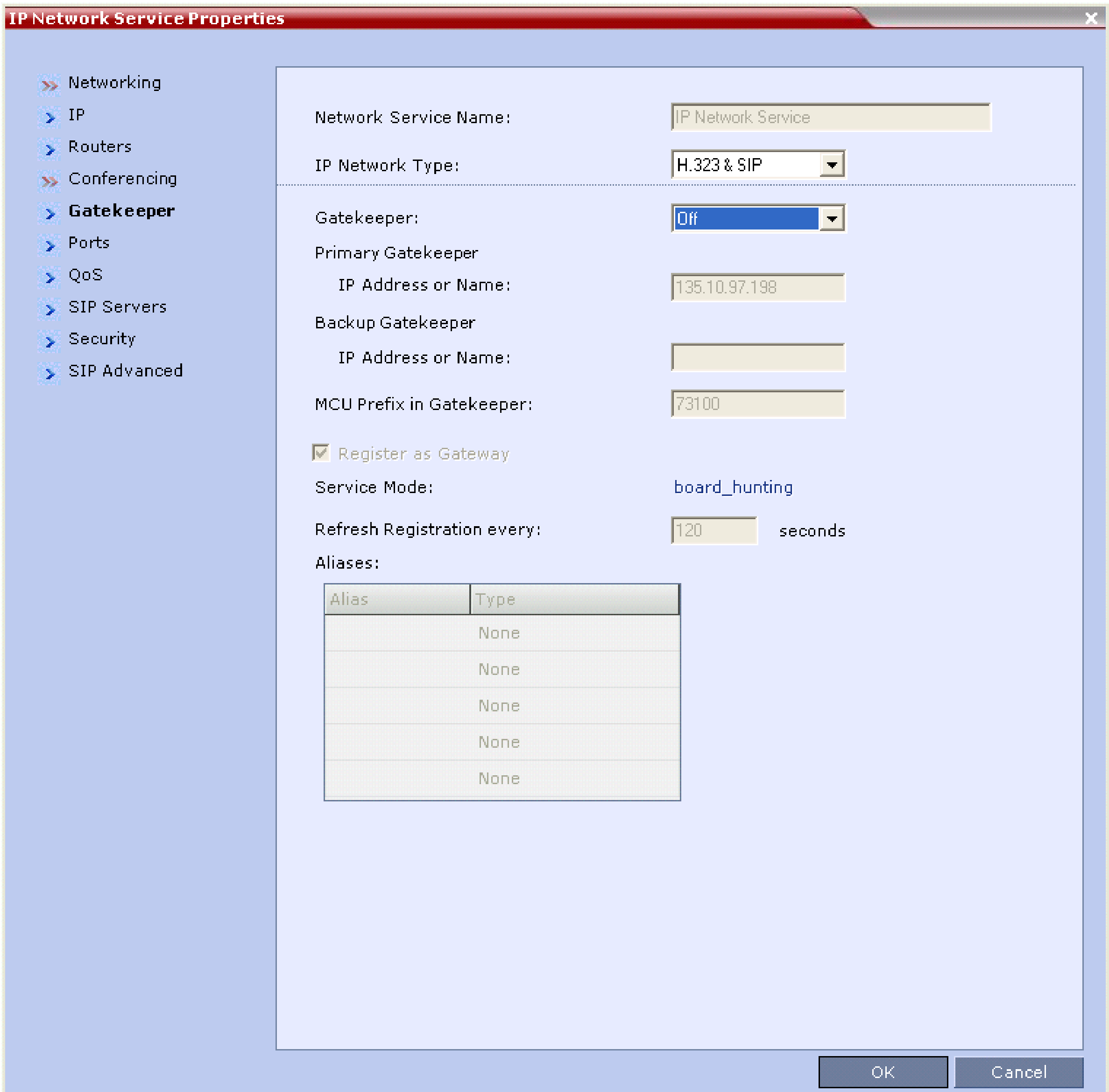

Figure 65: Gatekeeper section with Off selected

8.2. Configure Conference Profiles

To create a conference profile, in the homepage of RMX 4000 Web Client, click on the **Conference Profiles** icon under the **RMX Management** window. The **Conference Profiles** window appears on the right-hand side as shown in **Figure 66**.

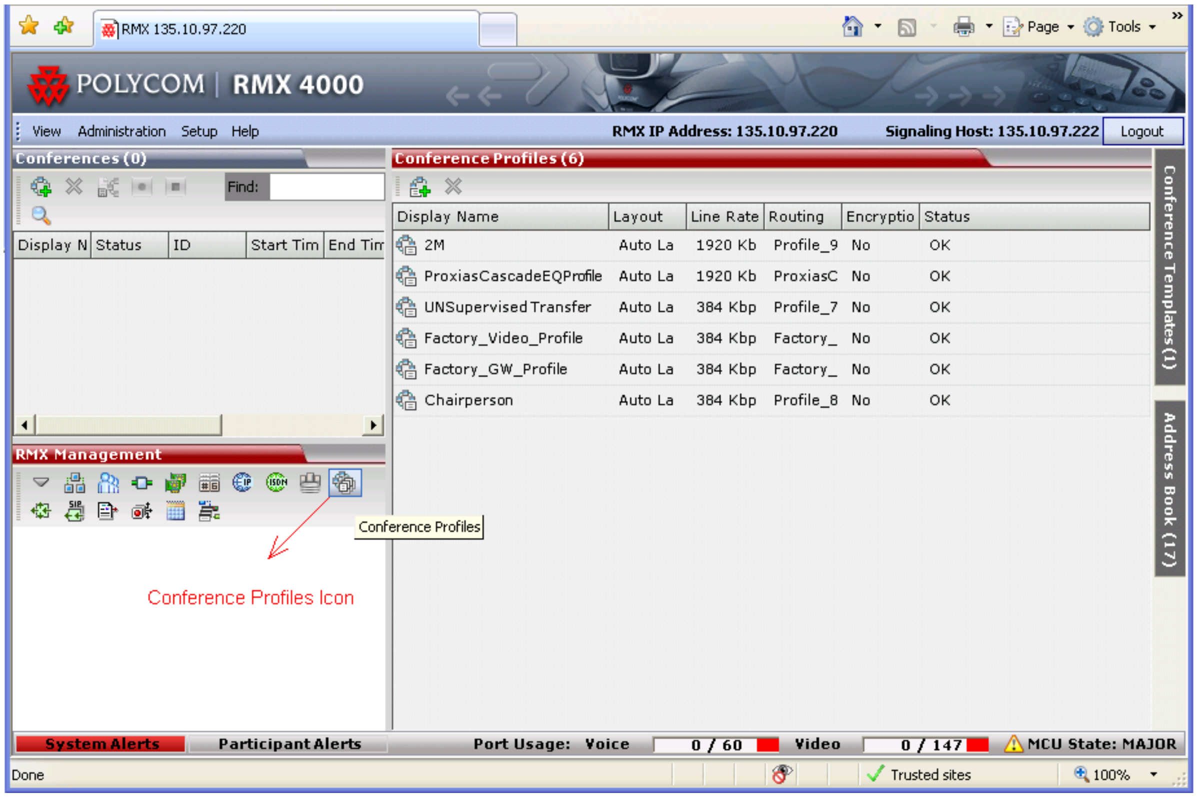

Figure 66: Conference Profiles Window

From the right-hand side of the **Conference Profiles** window, click on the **New Profile** icon to create a new conference profile as shown in **Figure 67** below.

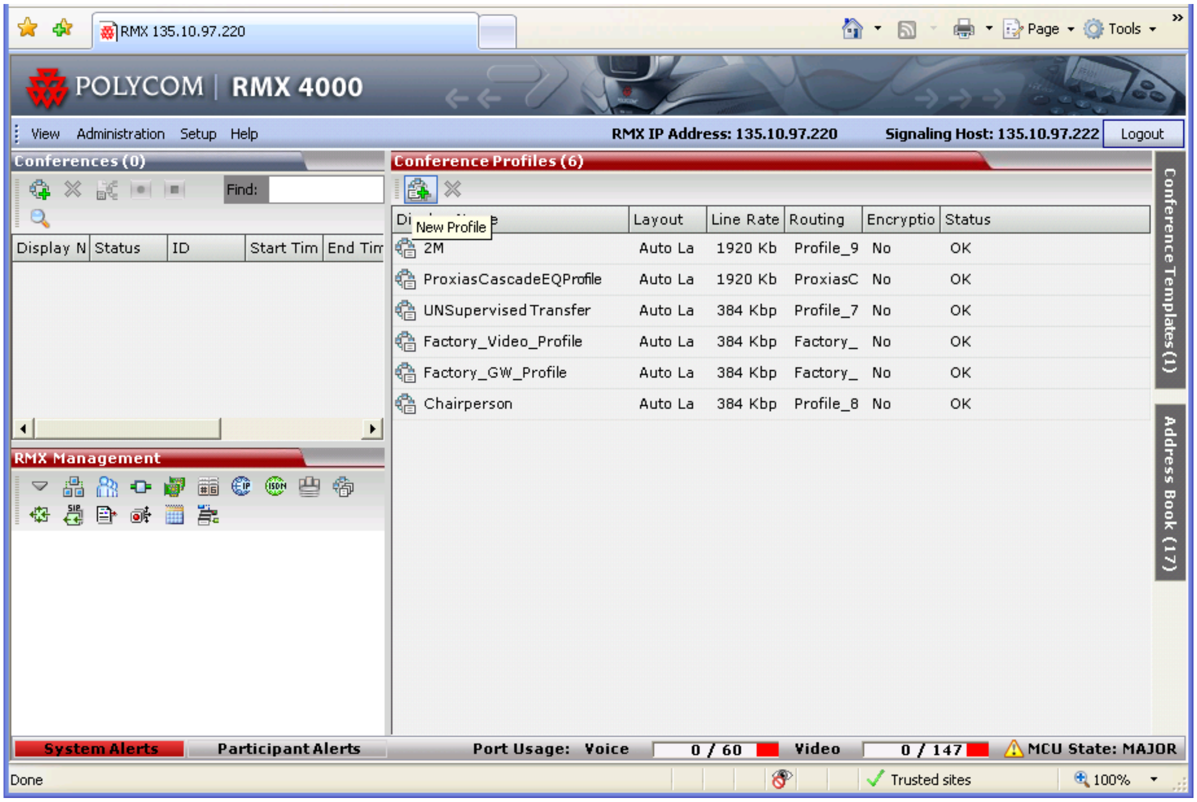

Figure 67: New Profile Icon of the Conference Profiles

The **New Profile** appears as shown in **Figure 68** below. Enter a name in the **Display Name** box for example *Video_Conf_Profile*, select the conference call bandwidth in the **Line Rate** field. The default is *384Kbps*, enter a name in the **Routing Name** box.

Click **OK** button to complete creating a new conference profile.

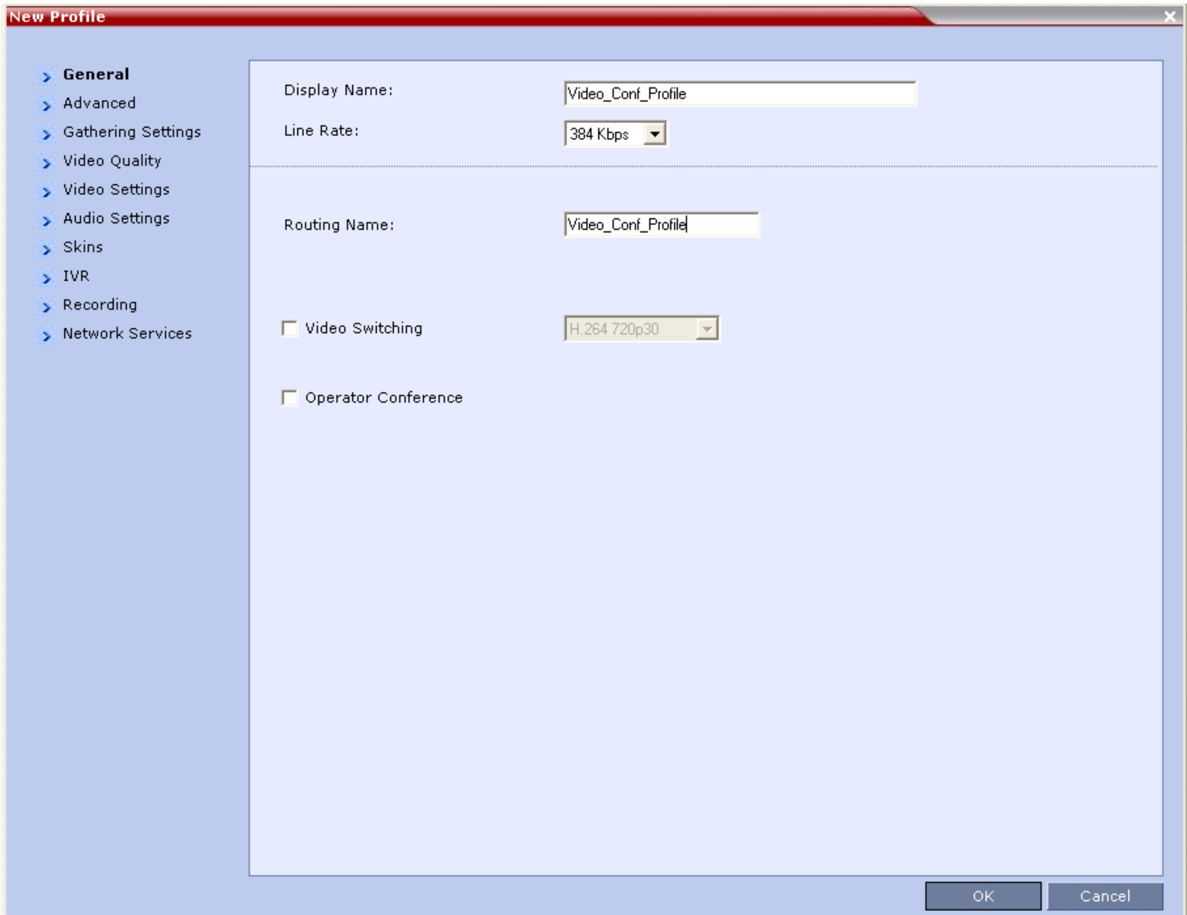

Figure 68: New Conference Profile window

Figure 69 below show the new conference profile **Video_Conf_Profile** has been created.

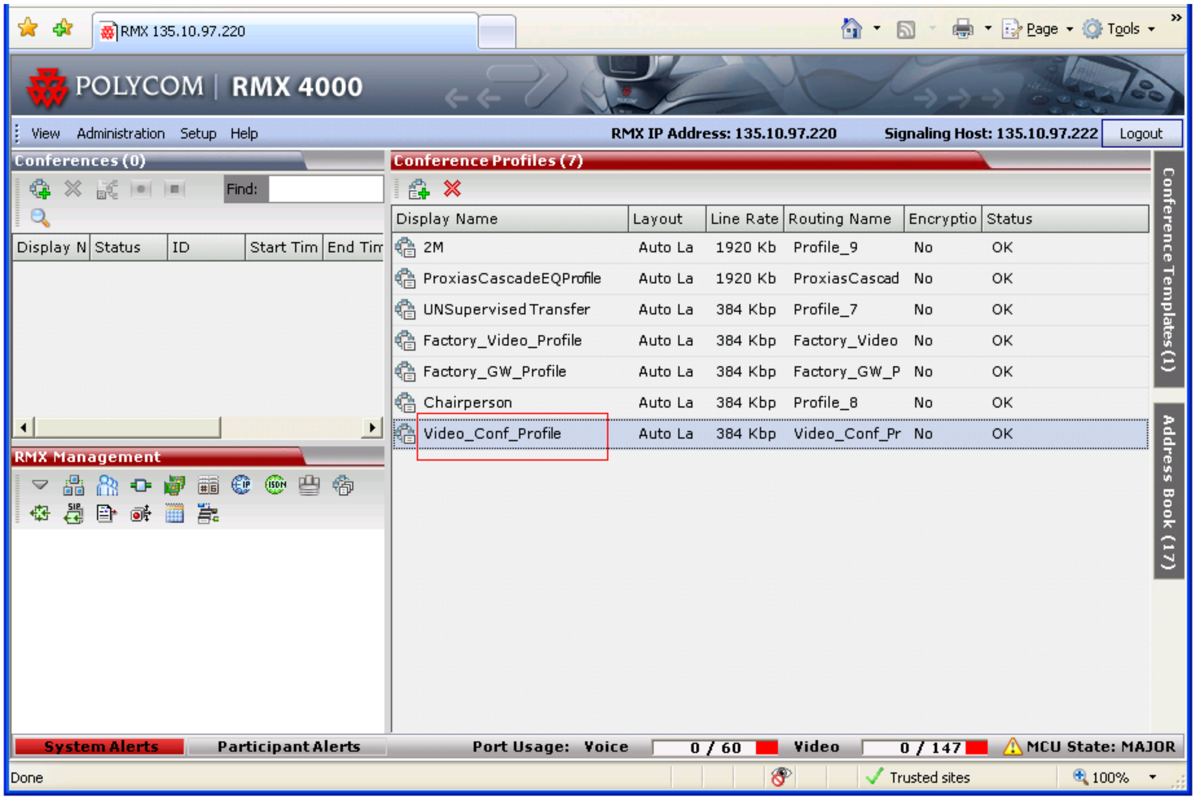

Figure 69: The Video_Conf_Profile new conference profile created

8.3. Configure Entry Queue

To create a new entry queue, in the homepage of RMX 4000 Web Client, click on the **Entry Queues** icon under the **RMX Management** window. The **Entry Queues** window appears on the right-hand side as shown in **Figure 70**.

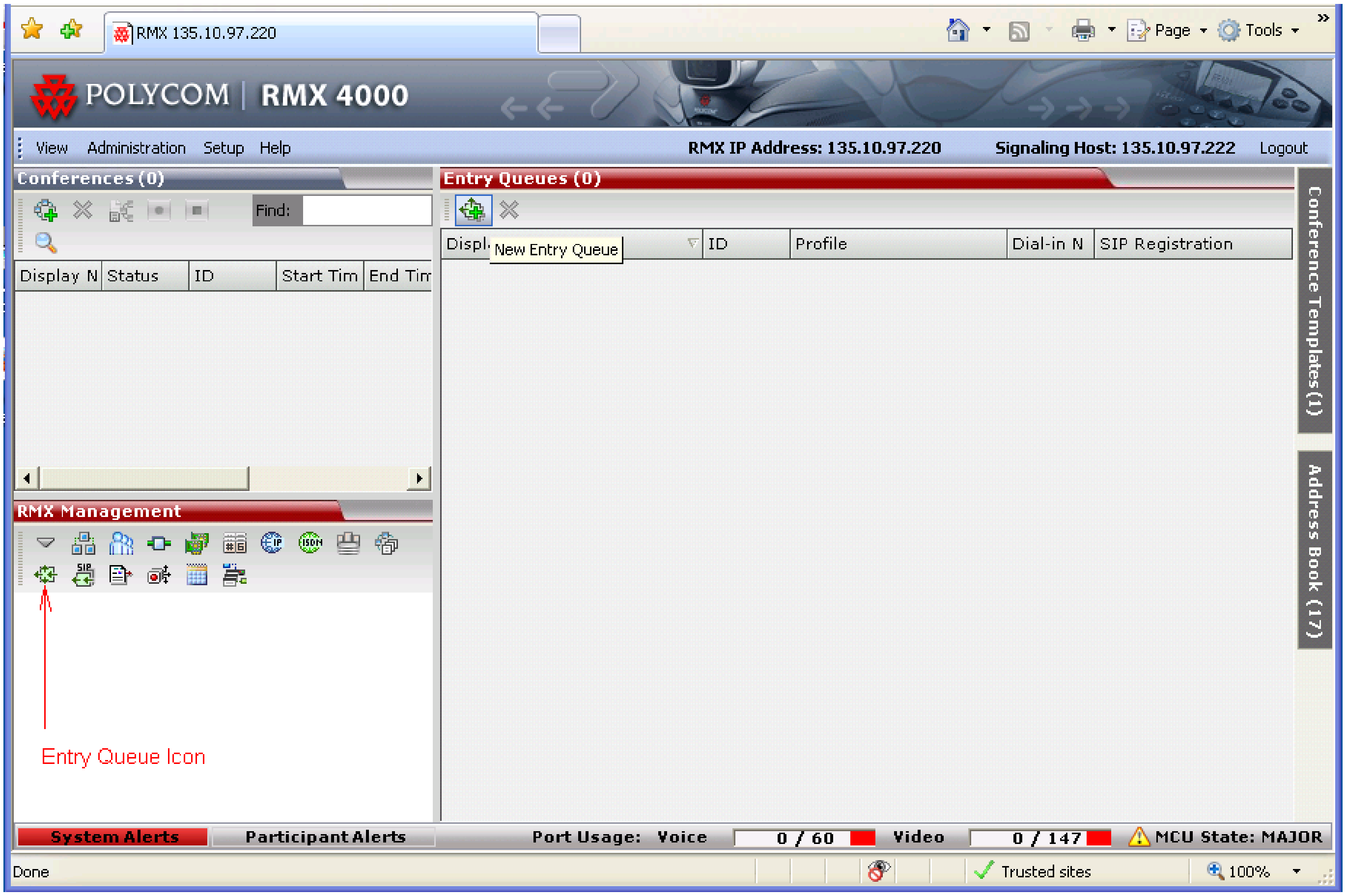

Figure 70: The Entry Queue Window

From the **Entry Queues** window, click on the **New Entry Queue** icon to create a new entry queue, the new entry queue window appears as shown in **Figure 71** below.

Enter the name *DefaultEQ* in the **Display Name** box, use the same name in the **Routing Name** box. In the **Profile** dropdown menu , select the profile *Video_Conf_Profile* that was created in **Section 8.2**, select *Entry Queue IVR Service* in the **Entry Queue IVR Service** dropdown menu, select *None* in the **Cascade** dropdown menu.

<u>Note</u>**:** The **Ad hoc** checkbox is optional. If this checkbox is checked, users can create their own conference room by calling into the RMX 4000, if this checkbox is un-checked user can only join in the existing conference rooms in the RMX 4000 system.

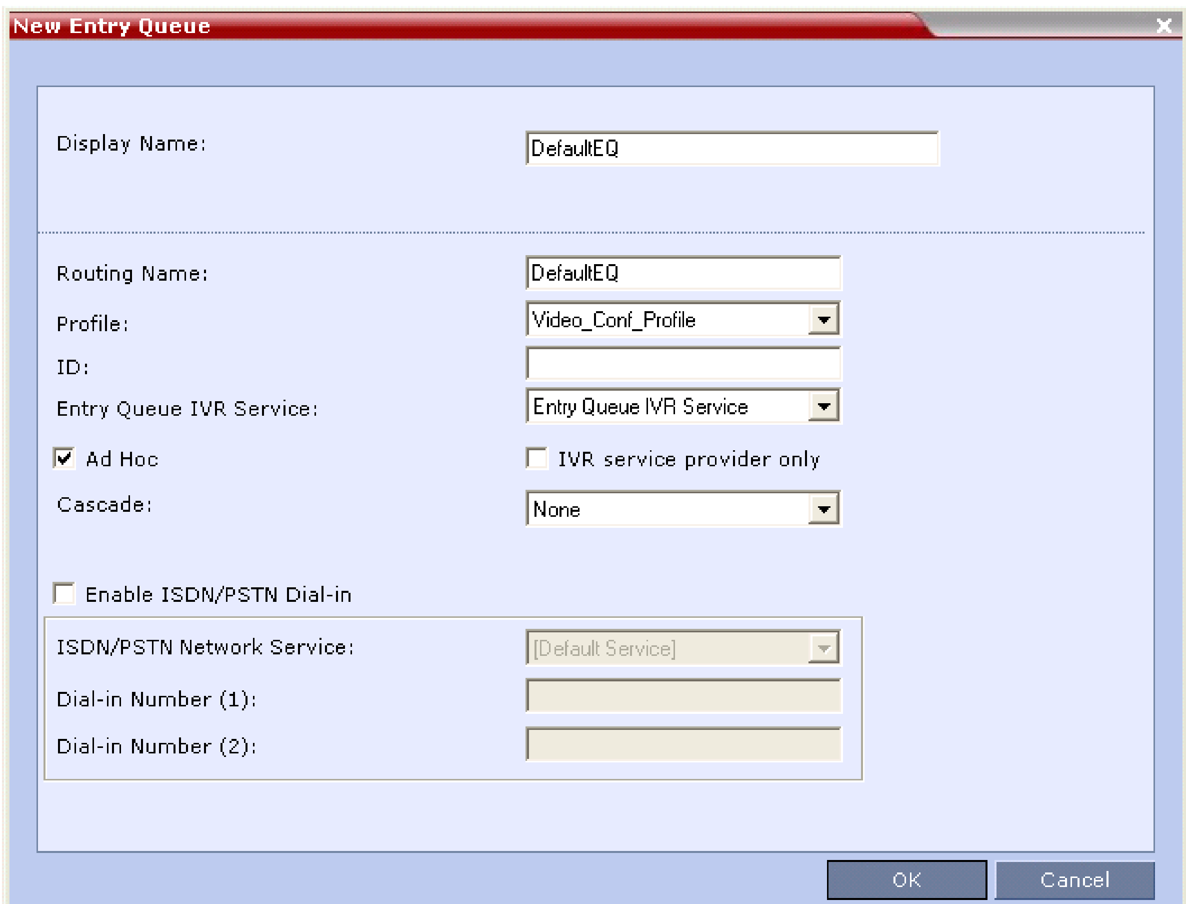

Figure 70: New Entry Queue Window

Click **OK** button to complete creating the new entry queue. **Figure 71** below shows the new entry queue *DefaultEQ* has been created.

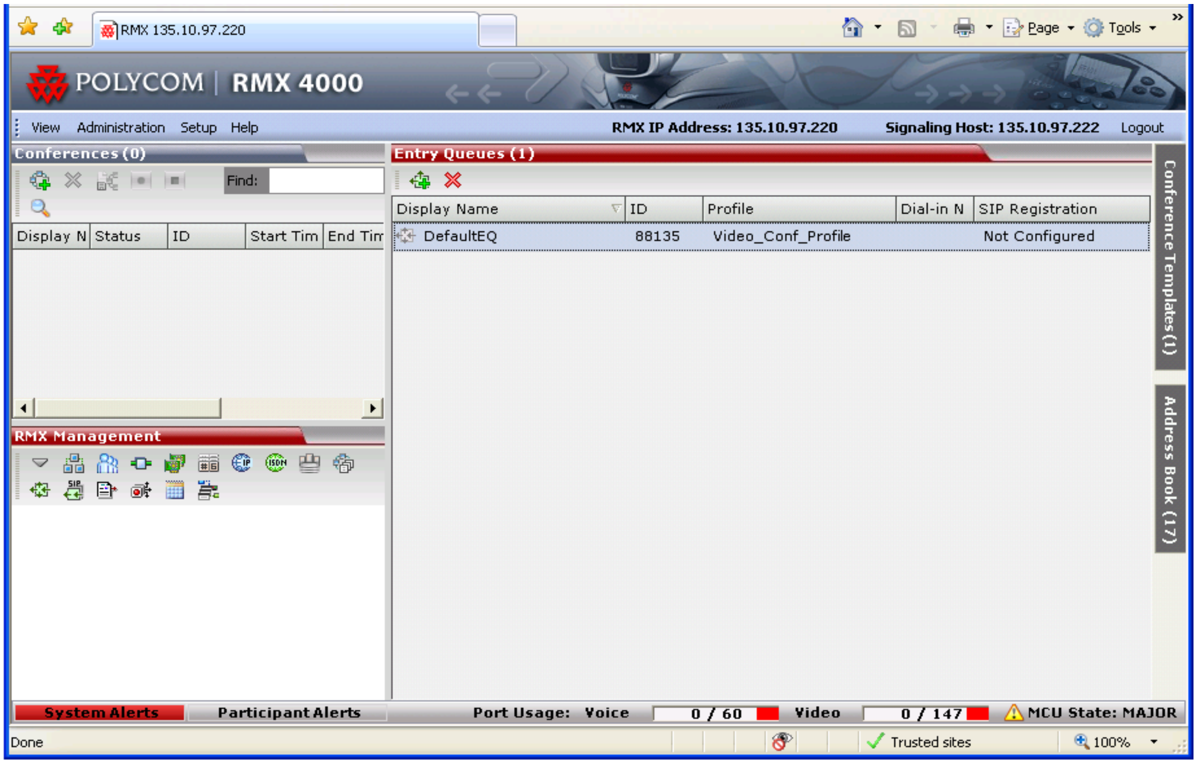

Figure 71: The DefaultEQ created

The new entry queue needs to be enabled, to do that right-click on the entry queue and select **Set Transit Entry Queue** as shown in **Figure 72**.

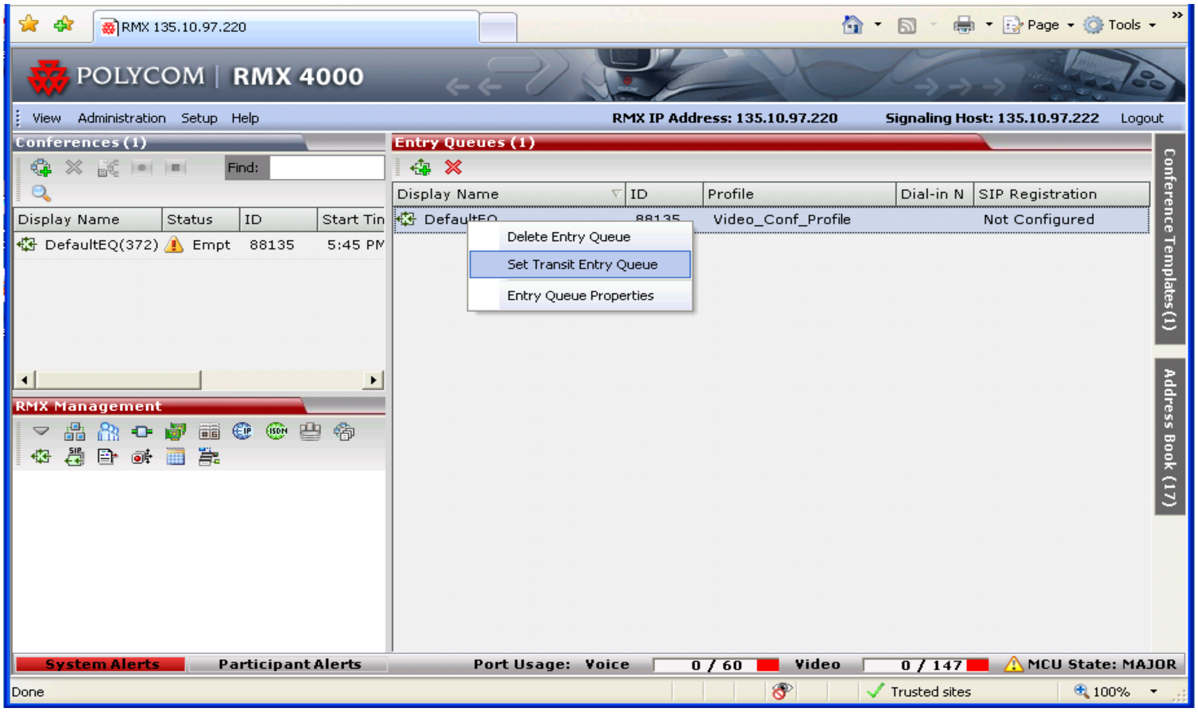

Figure 72: Set Transit for the DefaultEQ

8.4. Configure Meeting Room

To create a meeting room, in the homepage of RMX 4000 Web Client, click on the **Conference Rooms** icon under the **RMX Management** window. The **Meeting Rooms** window appears on the right-hand side as shown in **Figure 73**.

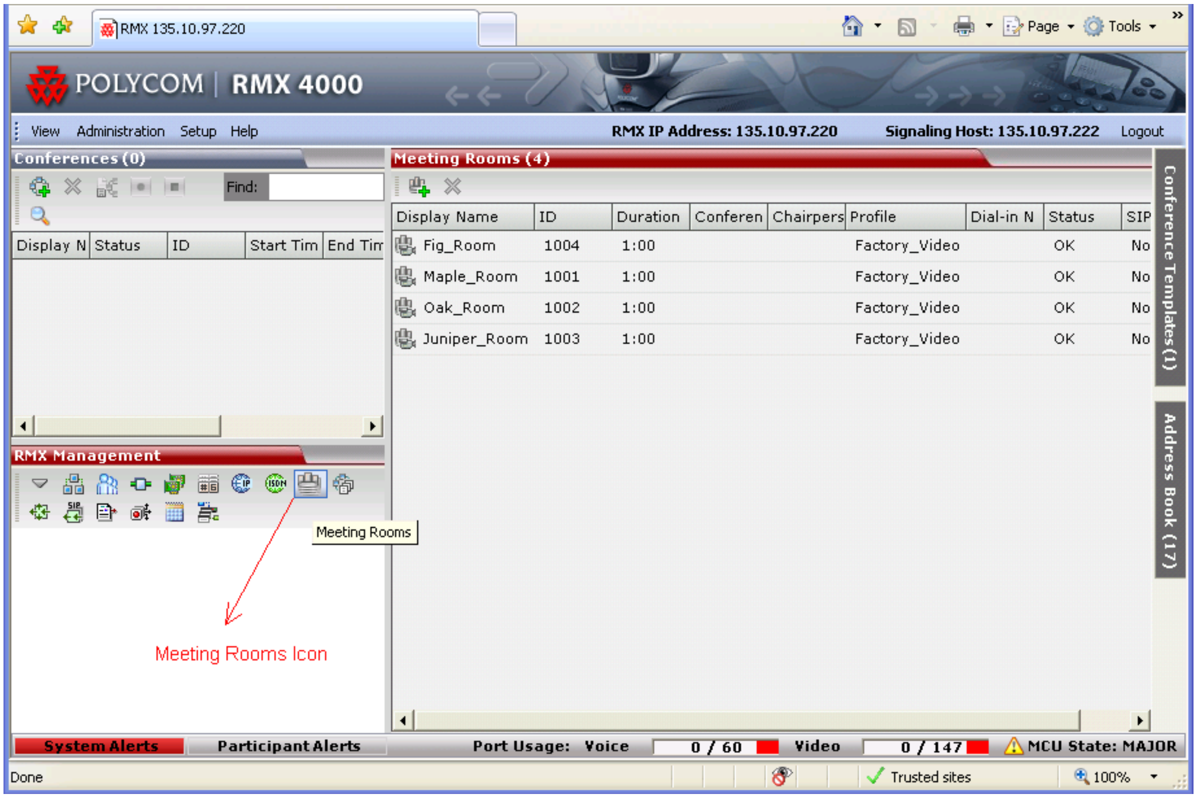

Figure 73: The Meeting Rooms window

From the **Meeting Rooms** window, click on the **New Meeting Rooms** icon to create a new meeting room, the new meeting window appears as shown in **Figure 74** below.

Enter a name in the **Display Name** box, for example *Bellevile_Room*, set duration time for this meeting room in the **Duration** field, enter a name in the **Routing Name** box, select conference profile in the **Profile** field, enter conference ID in the **ID** field, password in the **Conference Password**, password in the **Chairperson Password** field, and keep other fields as default.

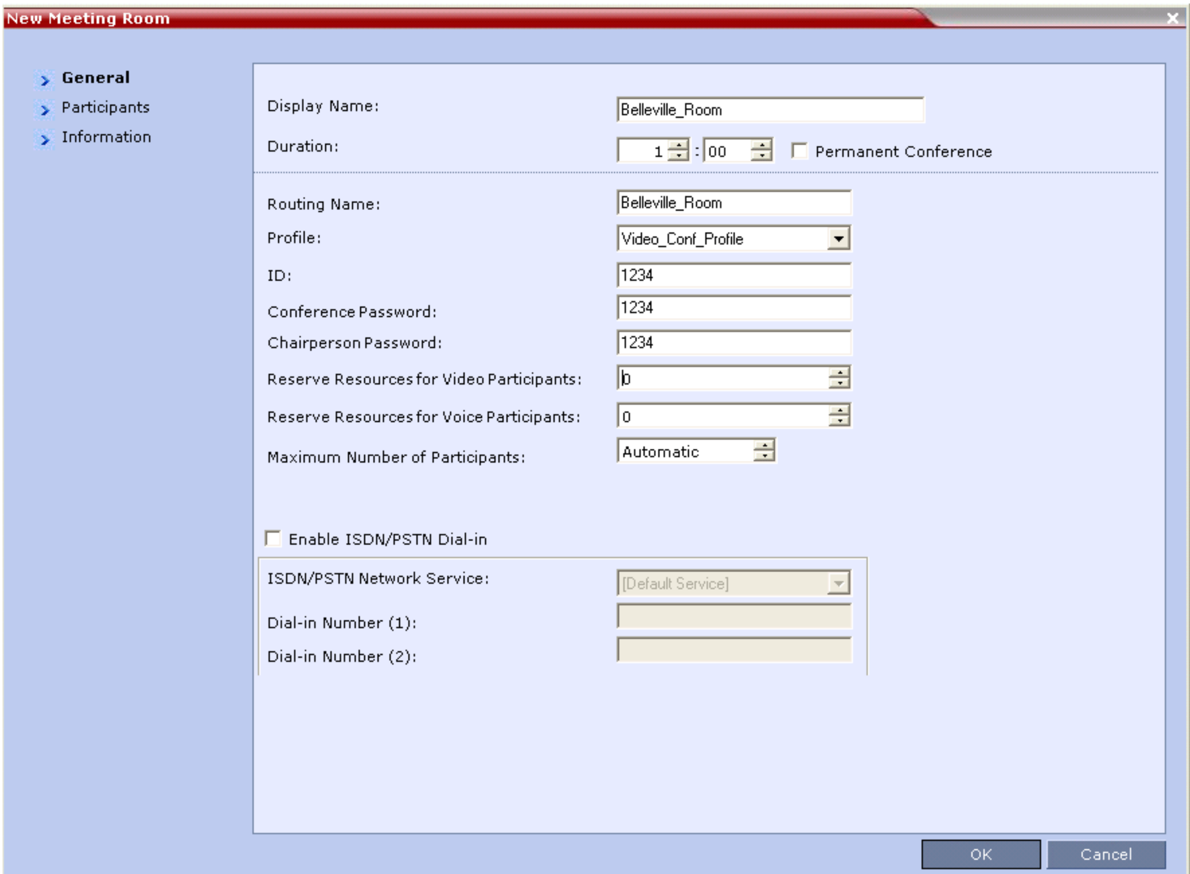

Figure 74: New Meeting Room window

Click **OK** button to complete adding the new meeting room, and the new meeting room name *Belleville_Room* has been created and appears in the **Meeting Rooms** window as shown in **Figure 75**.

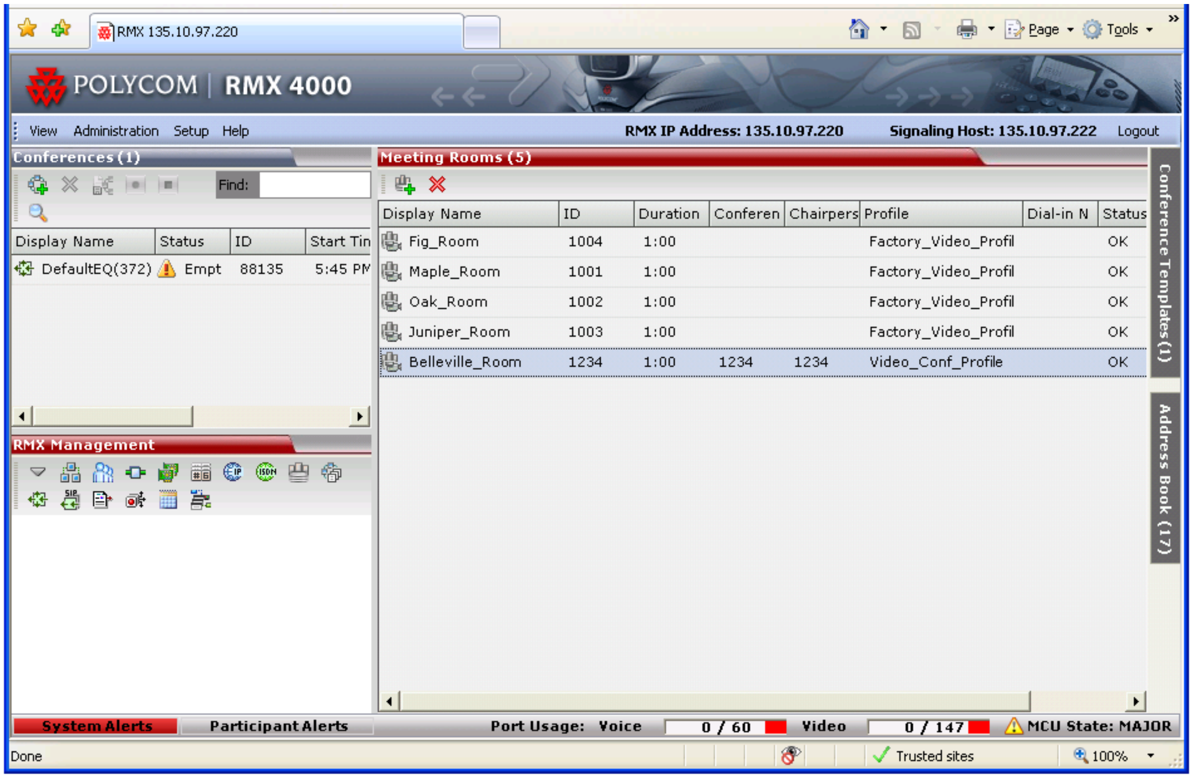

Figure 75: The Belleville_Room meeting room created

9. Verification Steps

The following are typical steps to verify the interoperability between the Polycom RMX 4000 system and Avaya Communication Server 1000 Release 7.5 and Avaya Aura® Session Manager.

- Provision 3 SIP user accounts for 3 Polycom HDX 8000 video endpoints and register them to the CS1000 SIP Line.
- Provision some IP Unistim phones in the CS1000 TPS server.
- From a conference room in the RMX 4000 invite two participants from the HDX endpoints to join the conference by dialing out their DNs.
- Accept the call on these HDX SIP endpoints, the video conference was established between both HDX endpoints with video and audio.
- From the third HDX endpoint dial in to the DN 73100 of RMX 4000 and verify the user was asked to enter a conference ID to join the conference. Enter ID of the conference above and join the video conference.
- From an Avaya IP Unistim phone dial in to the DN 58881 of CallPilot and then when prompted enter the DN 73100 of RMX 4000 to join the conference above.
- Check the video and audio on each SIP Video endpoint, it should be clear audio and sharp video, and only audio on the IP Unistim phone.

10. Conclusions

All of the executed test cases passed and met the objectives as outlined in **Section 2**. The Polycom RMX 4000 is considered compliant with Avaya Communication Server 1000 Release 7.5 and Avaya Aura® Session Manager Release 6.1.

11. Additional References

Product documentation for Avaya products may be found at:
https://support.avaya.com/css/Products/

Product documentation for Polycom RMX 4000 may be found at:
http://www.Polycom.com

Avaya Aura® Session Manager Documents:
[1] Administering Avaya Aura® Session Manager Release 6.1, Doc# 03-603324, Issue 2, Date November 2010.
[2] Administering Avaya Aura® System Manager Release 6.1, Date November 2010.

Avaya Communication Server 1000 Documents:
[3] Avaya Communication Installation and Commissioning, Doc# NN43041-310, Issue 05.04, Date May 2011.
[4] Avaya Communication Server 1000 Unified Communications Management Common Services Fundamentals, Doc # NN43001-116, Issue 05.11, Date June 2011.
[5] Avaya Communication Server 1000 Co-resident Call Server and Signaling Server Fundamentals, Doc # NN43001-509, Issue 03.02, Date June 2011.
[6] Avaya Communication Server 1000 Element Manager System Reference - Administration, Doc# NN43001-632, Issue 05.09, Date July 2011.

Avaya CallPilot® Messaging Documents:
[7] Avaya CallPilot® Desktop Messaging and My CallPilot Installation and Administration, Doc# NN44200-305, Issue 01.15, Date May 2011

Polycom RMX Documents:
[8] Polycom RMX 1500/2000/4000 Getting Started Guide, Version 7.2, May 2011, DOC2611A

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.