



## **Avaya Solution and Interoperability Test Lab**

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# **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 Server, Signaling Server SIP Gateway, and SIP Line server. Endpoints used were IP Unistim Phones, Digital and SIP Phones. The testing used the Avaya Aura<sup>®</sup> 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 CallPilot 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.

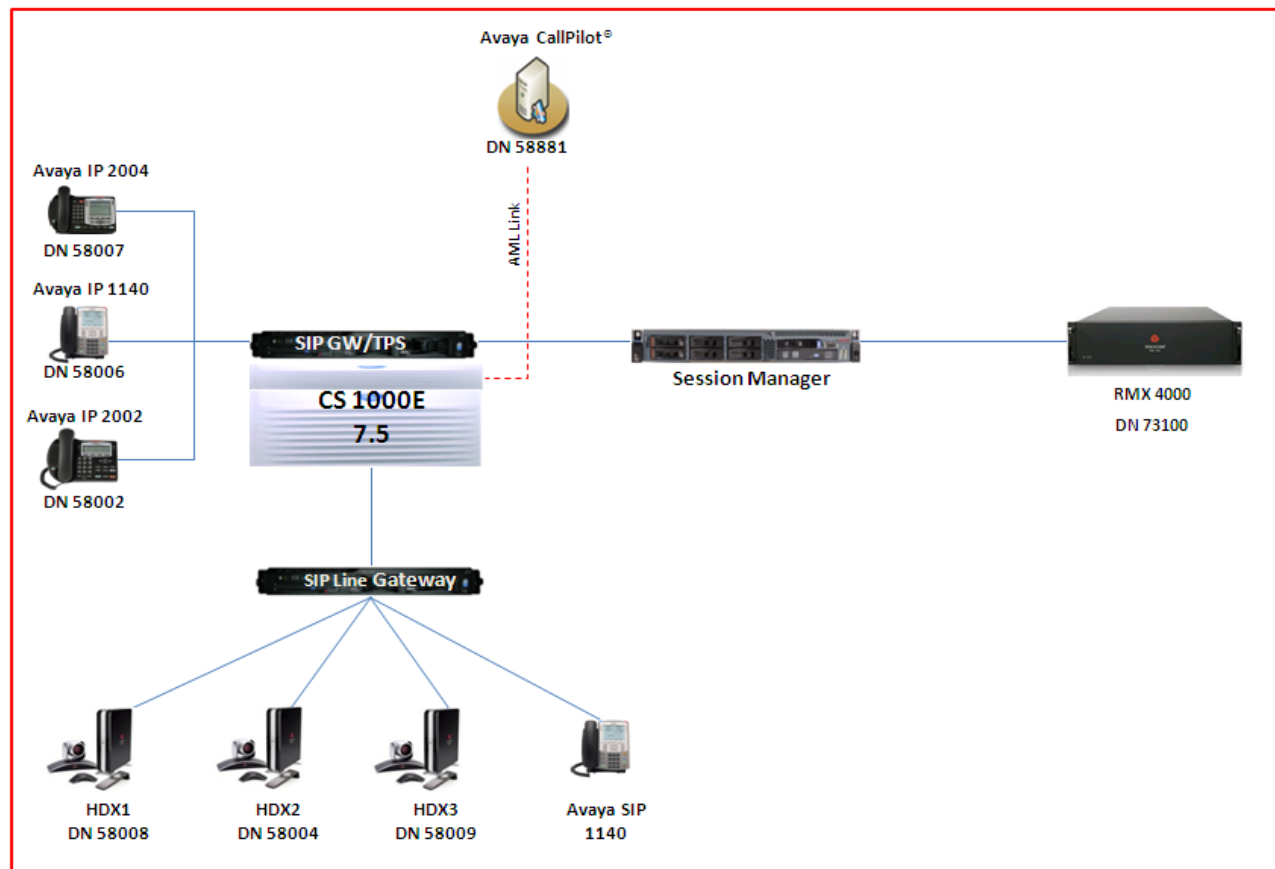
Observation: The CallPilot Messaging system doesn't support video calling, only audio calls therefore video calls made from a video endpoint to the CallPilot Messaging system which then continues dialling to the RMX 4000 is only able to join a conference with audio, even if the conference supports video. 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](http://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	0625C7F
Avaya IP 2004P2	0692D93
Avaya IP 2002P2	0604DCN
Avaya Digital M3905	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

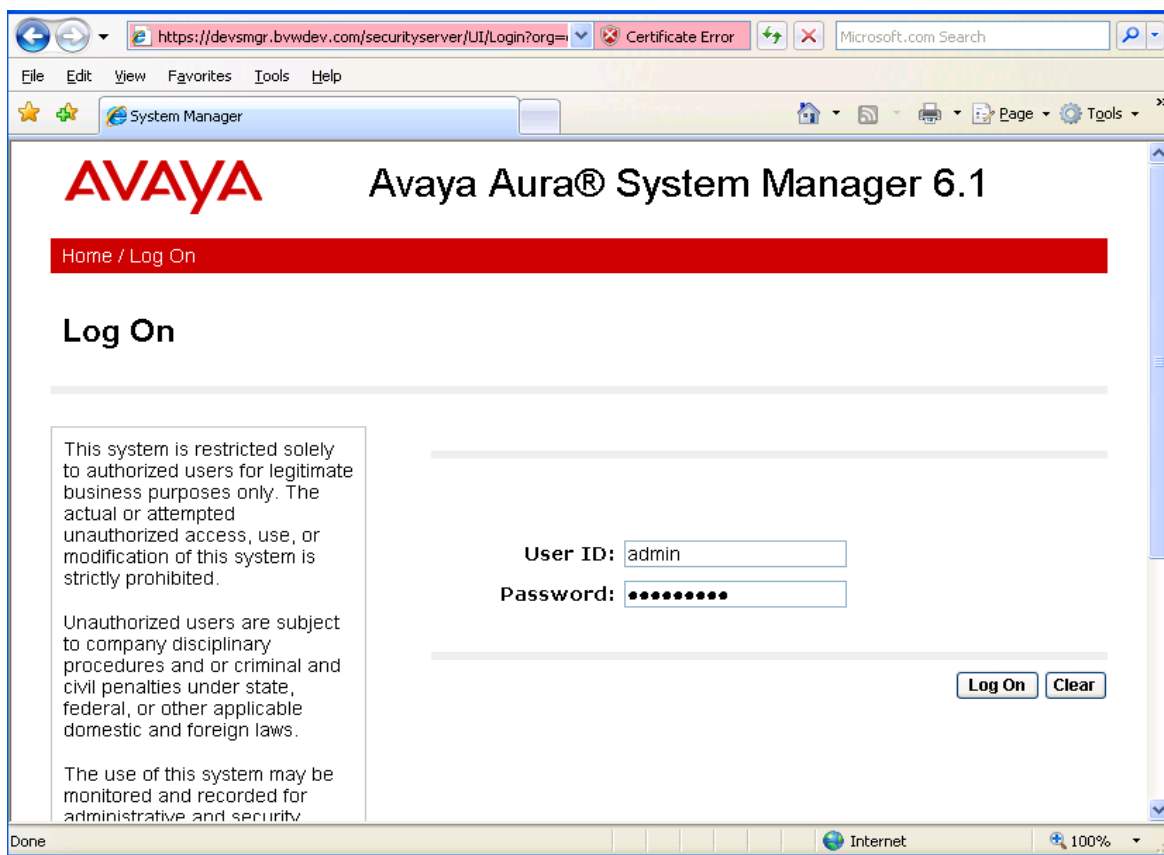
This document assumes that the Session Manager was properly installed and configured as per the product documents; this section provides the steps on how to provision the Session Manager working with the CS1000 and Polycom RMX 4000. For more information about how to install and configure Session Manager,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

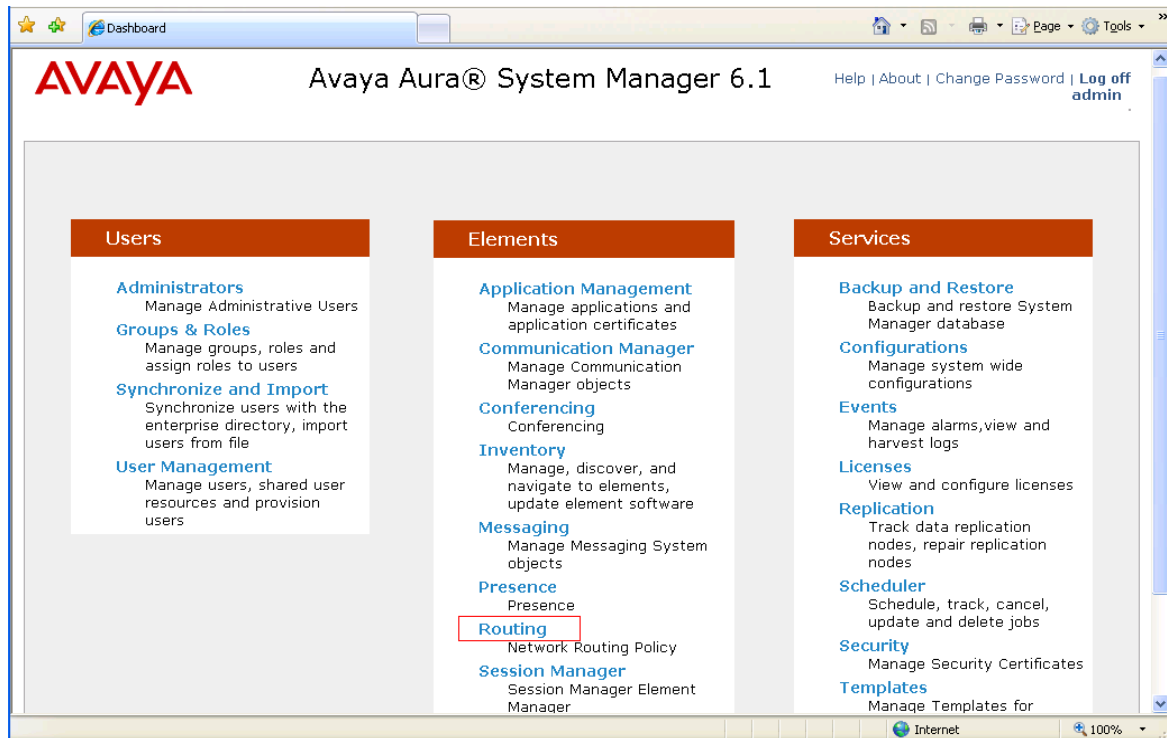
### 5.1. Configure SIP Domain

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.



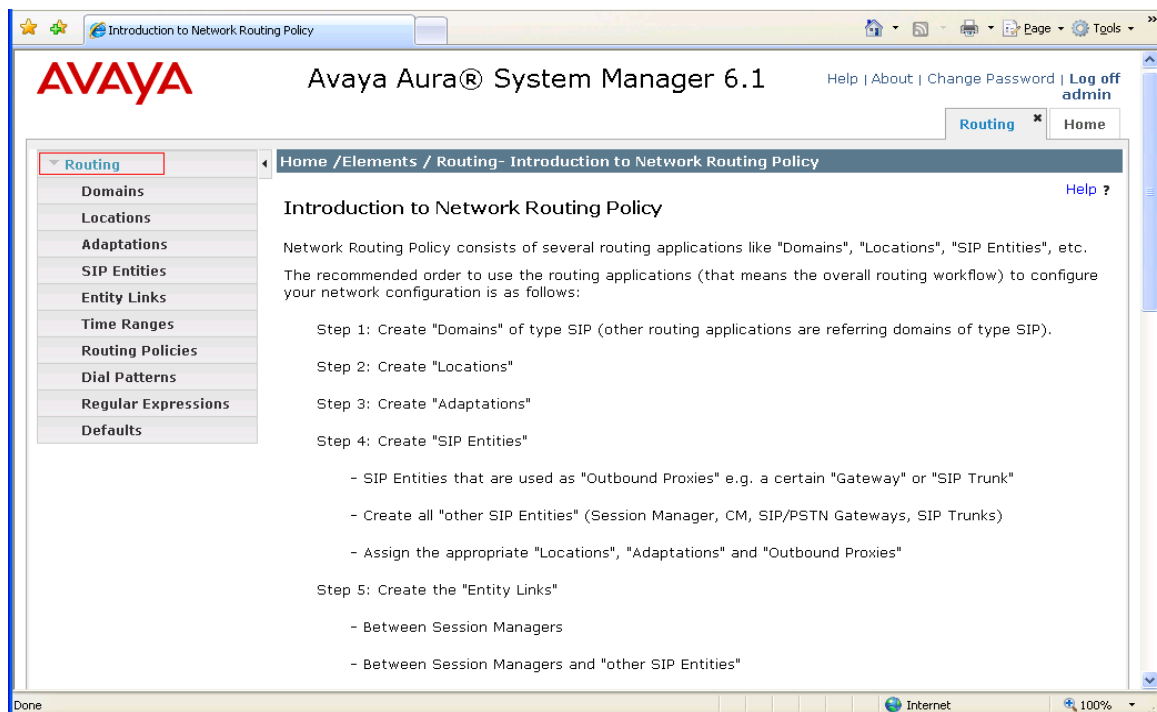
**Figure 2: Log On window of System Manger**

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



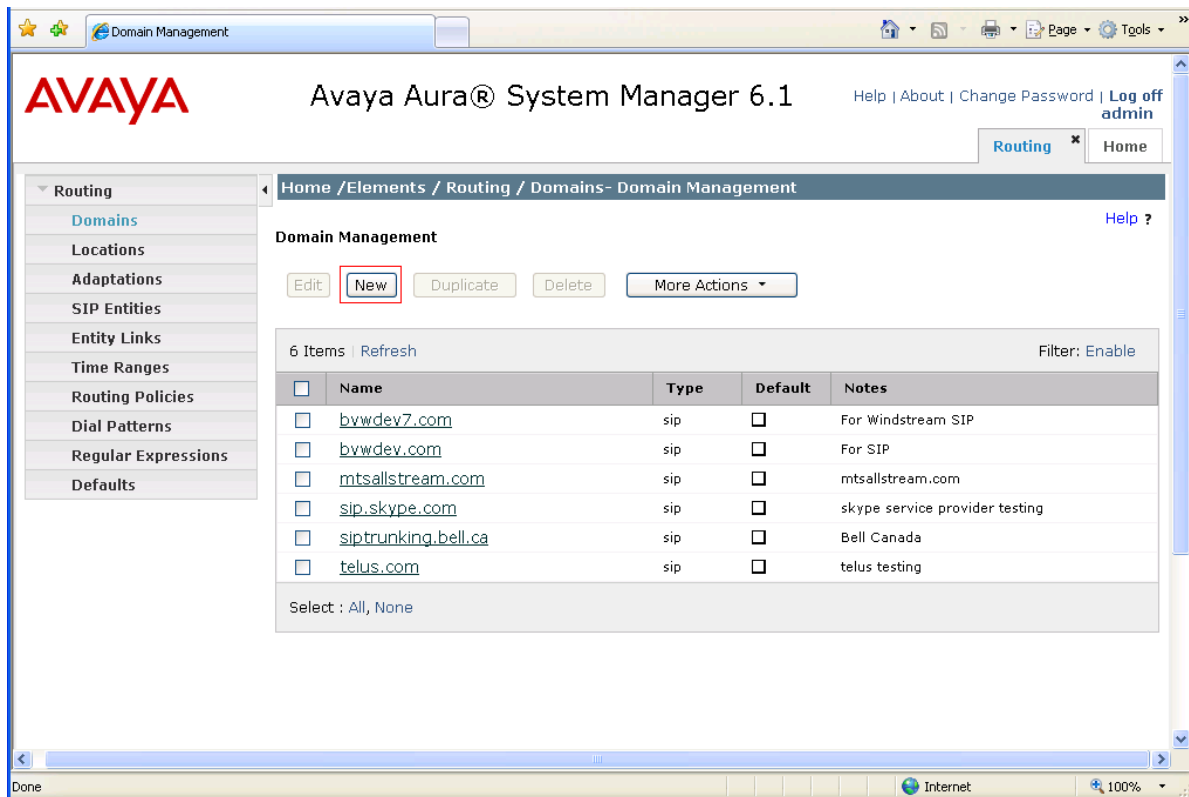
**Figure 3: Home page of System Manager**

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** **bwvdev.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 **bwvdev.com**.

The screenshot shows the Avaya Aura System Manager 6.1 web interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura® System Manager 6.1', and links for 'Help', 'About', 'Change Password', and 'Log off admin'. A breadcrumb trail indicates the current location: 'Home /Elements / Routing / Domains- Domain Management'. The left sidebar contains a tree view with 'Routing' expanded, showing sub-items like 'Domains', 'Locations', 'Adaptations', 'SIP Entities', 'Entity Links', 'Time Ranges', 'Routing Policies', 'Dial Patterns', 'Regular Expressions', and 'Defaults'. The main content area is titled 'Domain Management' and features a table with the following data:

Name	Type	Default	Notes
* bwvdev.com	sip	<input type="checkbox"/>	bwvdev.com sip domain

Below the table, there is a red asterisk and the text '\* Input Required'. At the bottom right of the form, the 'Commit' button is highlighted with a red rectangular box, next to a 'Cancel' button. The browser's status bar at the bottom shows 'Done' and 'Internet'.

**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.

The screenshot displays the 'Location Details' configuration page. The left sidebar contains a navigation menu with the following items: Routing, Domains, Locations (highlighted), Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Location Details' and includes a 'Help ?' link, 'Commit', and 'Cancel' buttons. A message states: 'Call Admission Control has been set to ignore SDP. All calls will be counted using the Default Audio Bandwidth. See Session Manager -> Session Manager Administration -> Global Setting'. The 'General' section contains a red asterisk next to the 'Name' field, which is filled with 'Belleville, Ont, Ca', and an empty 'Notes' field. The 'Overall Managed Bandwidth' section shows 'Managed Bandwidth Units' as a dropdown menu set to 'Kbit/sec' and 'Total Bandwidth' as a text box containing '1000000'. The 'Per-Call Bandwidth Parameters' section has a red asterisk next to the 'Default Audio Bandwidth' field, which is set to '100' with a 'Kbit/sec' dropdown. The 'Location Pattern' section includes 'Add' and 'Remove' buttons. At the bottom, there is a status bar with '0 Items', a 'Refresh' button, and a 'Filter: Enable' dropdown. The browser's address bar shows 'Home / Elements / Routing / Locations - Location Details'.

**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 *c ppm1* 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.

The screenshot displays the 'SIP Entity Details' configuration window. On the left, a sidebar lists various configuration categories: Routing, Domains, Locations, Adaptations, SIP Entities (selected), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'SIP Entity Details' and has a 'General' tab selected. The 'General' tab contains the following fields and values:

- Name:** c ppm1
- \* FQDN or IP Address:** 135.10.97.130
- Type:** Other
- Notes:** CS1000 SIP Entity
- Adaptation:** (empty dropdown)
- Location:** Belleville, Ont, Ca
- Time Zone:** America/New\_York
- Override Port & Transport with DNS SRV:** ☐
- \* SIP Timer B/F (in seconds):** 4
- Credential name:** (empty text box)
- Call Detail Recording:** none
- SIP Link Monitoring:** Use Session Manager Configuration

At the bottom of the window, there is a status bar showing 'Done' and 'Internet' connectivity.

**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**.

The screenshot shows the 'SIP Entity Details' configuration window. The left sidebar lists navigation options: Routing, Domains, Locations, Adaptations, SIP Entities (selected), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'SIP Entity Details' and has a 'General' tab selected. The 'General' tab contains the following fields:

- Name:** Polycom\_RMX1
- FQDN or IP Address:** 135.10.97.222
- Type:** Session Manager
- Notes:** For Polycom RMX1
- Location:** Belleville,Ont,Ca
- Outbound Proxy:** DevASM
- Time Zone:** America/New\_York
- Credential name:** (empty)

Below the 'General' tab is the 'SIP Link Monitoring' section with a dropdown menu set to 'Use Session Manager Configuration'. At the bottom, there is an 'Entity Links' section with 'Add' and 'Remove' buttons.

**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 ccppm1** as defined in the **Section 5.3** and **Port 5060** as shown in **Figure 10**.

The screenshot shows the 'Entity Links' configuration window. The left sidebar is the same as in Figure 9, but 'Entity Links' is now selected. The main content area is titled 'Entity Links' and has a 'Commit' and 'Cancel' button. Below this is a table with 1 item. The table has the following columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, and Port. The data row is as follows:

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port
* TCP_Link	* DevASM	TCP	* 5060	* ccppm1	* 5060

Below the table is a section labeled '\* Input Required' with 'Commit' and 'Cancel' buttons.

**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**.

The screenshot shows a web browser window with the title 'Entity Links'. The address bar shows 'Home /Elements / Routing / Entity Links- Entity Links'. The page has a sidebar on the left with a tree view containing: Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links (selected), Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Entity Links' and has 'Commit' and 'Cancel' buttons. Below this is a table with the following data:

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port
* TCP_Link	* DevASM	TCP	* 5060	* Polycom_RMX1	* 5060

Below the table, there is a message '\* Input Required' and another set of 'Commit' and 'Cancel' buttons. The browser's status bar at the bottom shows 'Internet' and '100%' zoom.

**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 *c ppm1* (not shown) as defined in **Section 5.3**. Click on the **Commit** button to save the new route.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing Policy Details

Routing Policy Details

General

\* Name:

Disabled: ☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
c ppm1	135.10.97.130	Other	

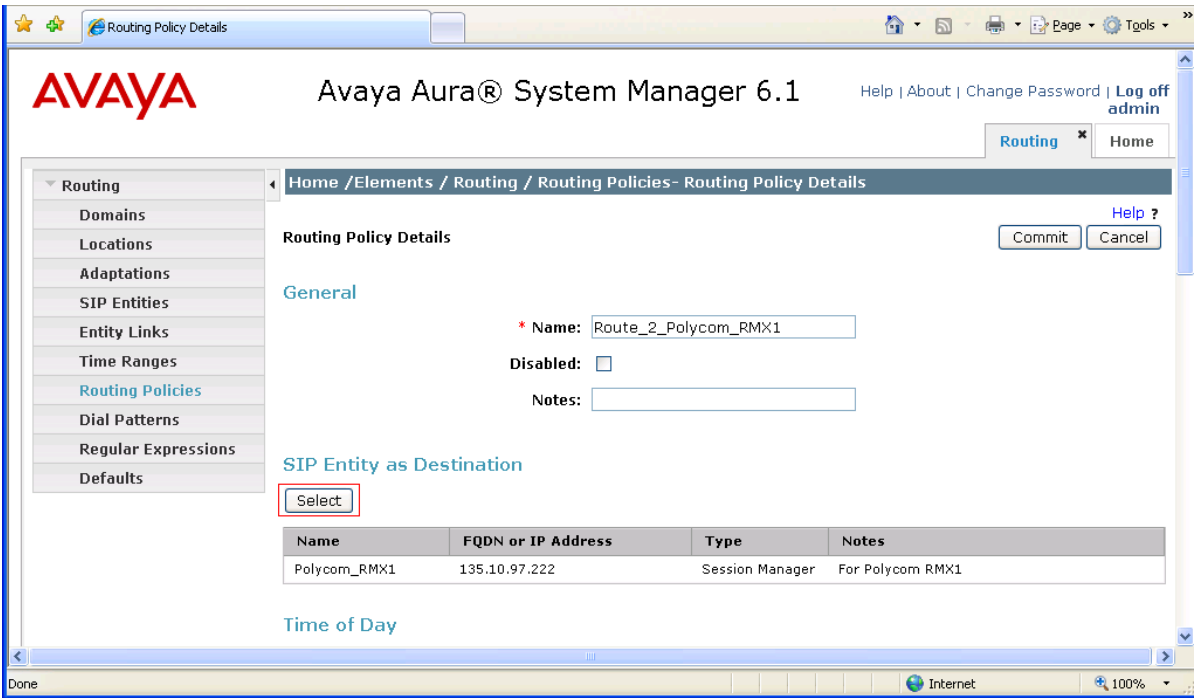
Time of Day

Add Remove View Gaps/Overlaps

Commit Cancel

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\_CSIK* as defined in **Section 5.3**. Click on the **Commit** button to save the new dial pattern.

**Dial Pattern Details**

**General**

\* Pattern: 58

\* Min: 5

\* Max: 5

Emergency Call: ☐

SIP Domain: bvwdev.com

Notes: Dial Pattern for CS1000 Routing

**Originating Locations and Routing Policies**

Add Remove

1 Item Refresh Filter: Enable

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Belleville, Ont, Ca		Routing_2_CSIK	0	<input type="checkbox"/>	cppm1	Routing to CS1000 cppm1

Select : All, None

**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**.

**Dial Pattern Details**

General

\* Pattern: 731

\* Min: 5

\* Max: 11

Emergency Call: ☐

SIP Domain: bvwddev.com

Notes: Dial pattern for Polycom RMX1

Originating Locations and Routing Policies

Add Remove

1 Item Refresh Filter: Enable

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Belleville, Ont, Ca		Routing 2 Polycom RMX1	0	<input type="checkbox"/>	Polycom_RMX1	Routing to Polycom_RMX1

Select: All, None

**Figure 15: Adding new dial pattern for RMX 4000 Route**

## 6. Configure Avaya Communication Server 1000

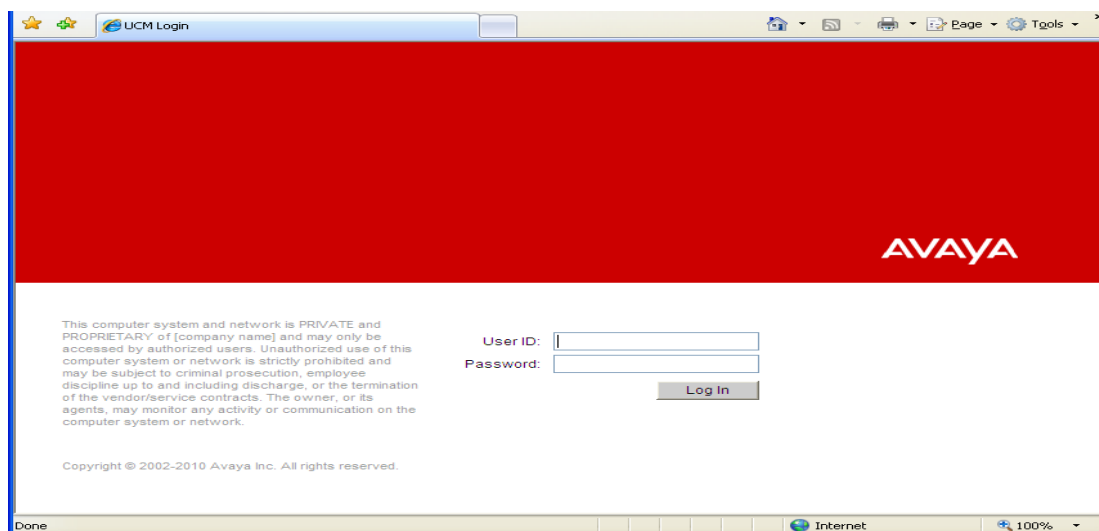
This document assumes that the Avaya Communication Sever 1000 system was properly installed and configured as per the product documents, this section provides the steps on how to provision the CS1000 working with the Session Manager and Polycom RMX 4000. For more information about how to install and 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

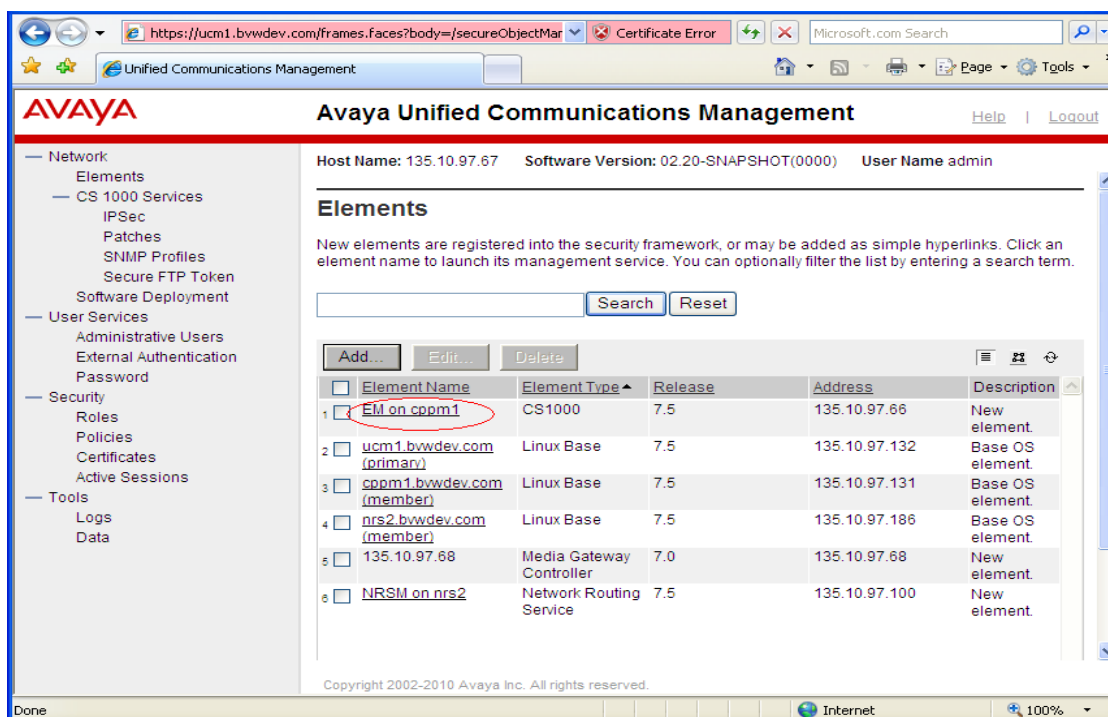
To register the CS1000 SIP Signaling Gateway to the Session Manager, follow the procedures below:  
Log in to the Unified Communication Management (UCM) managing the CS1000 system that needs to be configured, the UCM login window is shown as **Figure 16**.



**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**.

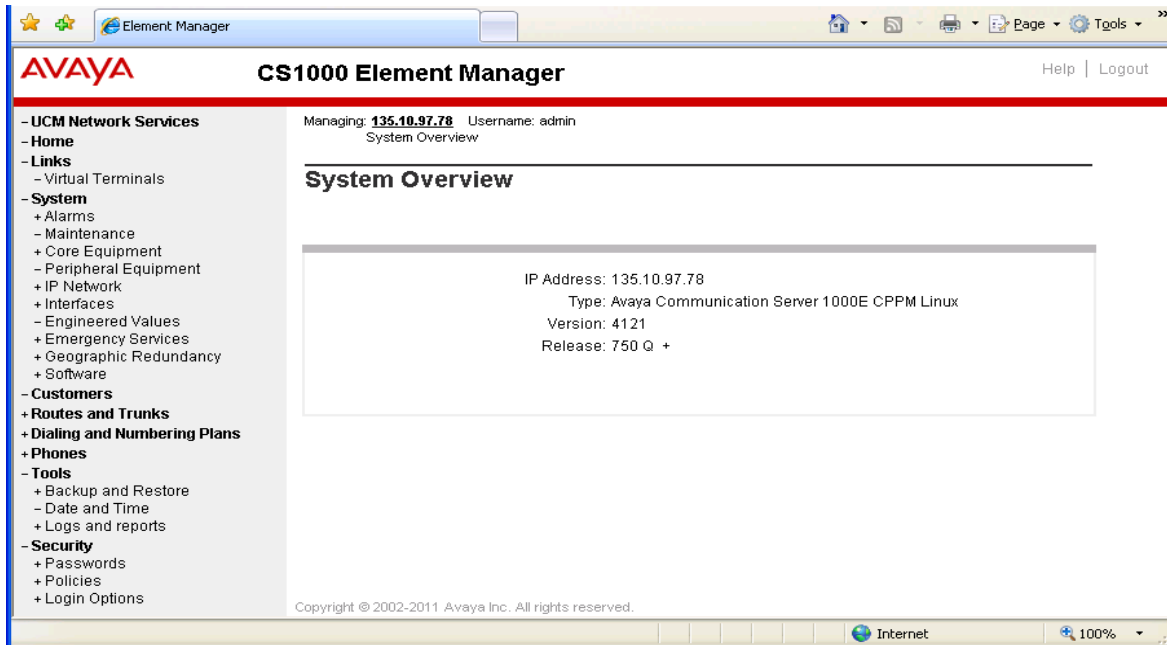
Click on the **Element Name** link (in this sample it is *EM on cppm1*) that manages the CS1000 system,.





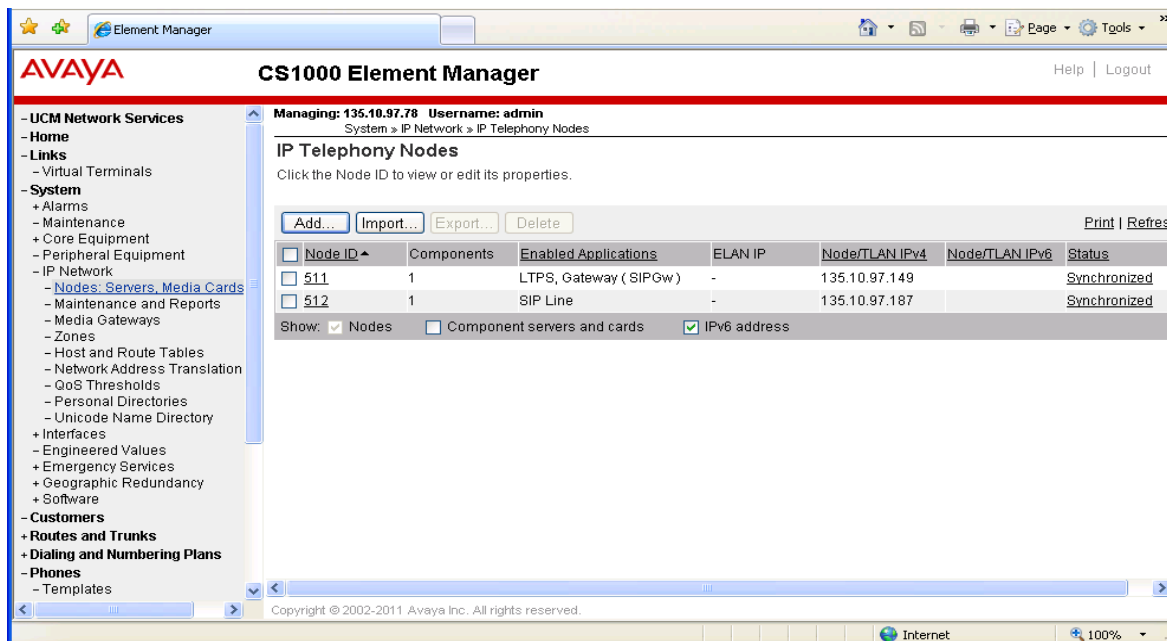
**Figure 17: UCM Home Page**

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



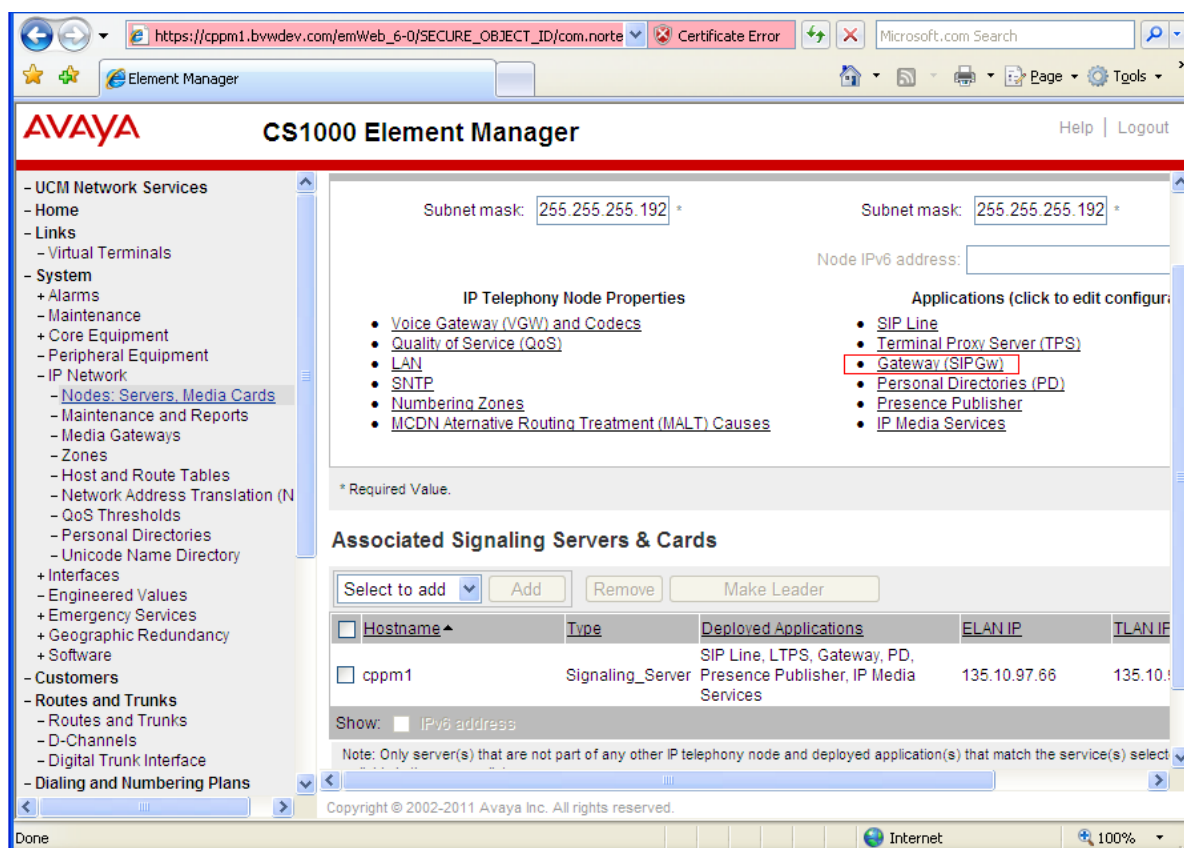
**Figure 18: CS1000 Element Manager Home page**

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.



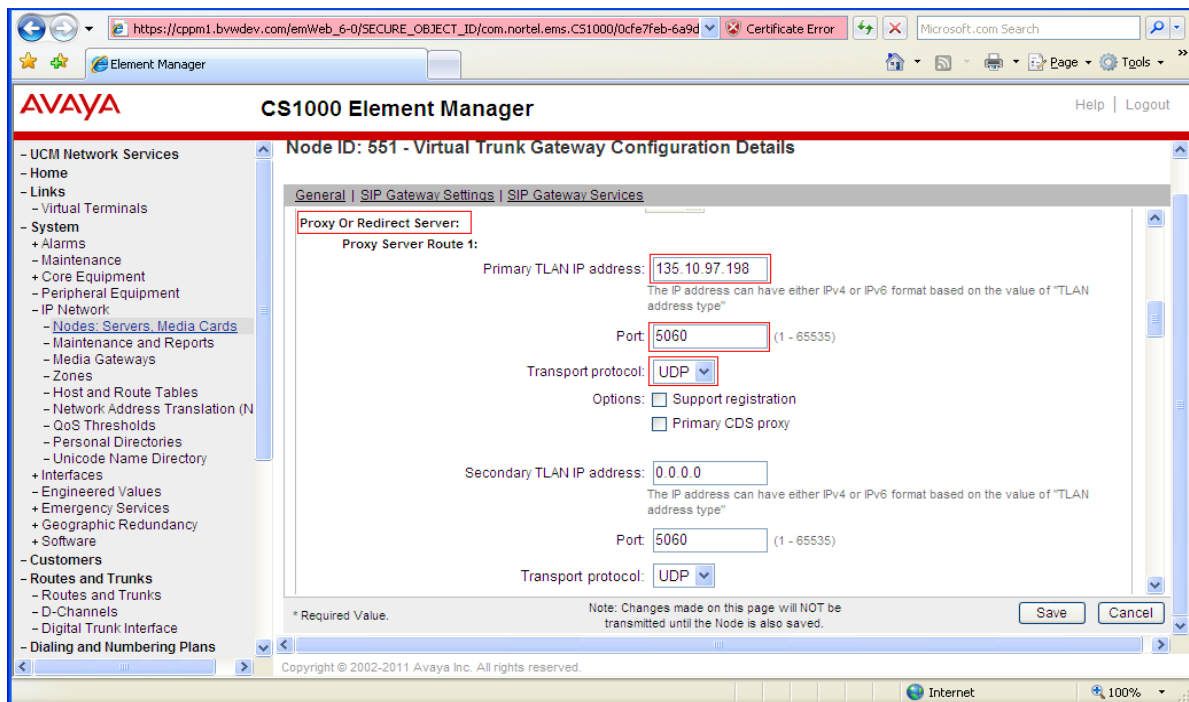
**Figure 20: IP Telephony Node Detail Page**

The **Node ID: 551 – Virtual Trunk Gateway Configuration Details** appears. In the **General** section, enter the domain *bwvdev.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**.

The screenshot shows the Avaya CS1000 Element Manager web interface. The browser address bar displays `https://cppm1.bvwdev.com/emWeb_6-0/SECURE_OBJECT_ID/com.norte`. The page title is "AVAYA CS1000 Element Manager". The left sidebar contains a navigation tree with categories like "UCM Network Services", "System", "Interfaces", "Customers", "Routes and Trunks", and "Dialing and Numbering Plans". The main content area is titled "Node ID: 551 - Virtual Trunk Gateway Configuration Details". It includes a breadcrumb trail: "System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration". Below the title, there are tabs for "General", "SIP Gateway Settings", and "SIP Gateway Services". The "General" tab is selected. It contains a checkbox for "Vtrk gateway application" which is checked, with the text "Enable gateway service on this node". Below this, the "General" section has several input fields: "Vtrk gateway application" (a dropdown menu showing "SIP Gateway (SIPGw)"), "SIP domain name" (text box with "bwvdev.com"), "Local SIP port" (text box with "5060" and a range indicator "(1 - 65535)"), "Gateway endpoint name" (text box with "cppm1"), "Gateway password" (text box), "Application node ID" (text box with "551" and a range indicator "(0-9999)"), and "Enable failsafe NRS" (checkbox). At the bottom of the "General" section, there is a "SIP ANAT" section with radio buttons for "IPv4" (selected) and "IPv6". To the right of the "General" section is the "Virtual Trunk Network Health Monitor" section, which includes a checkbox for "Monitor IP addresses (listed below)" and a text box for "Monitor IP:". Below these is a "Monitor addresses:" section with a large text area. The footer of the page shows "Copyright © 2002-2011 Avaya Inc. All rights reserved." and a status bar with "Done" and "Internet" icons.

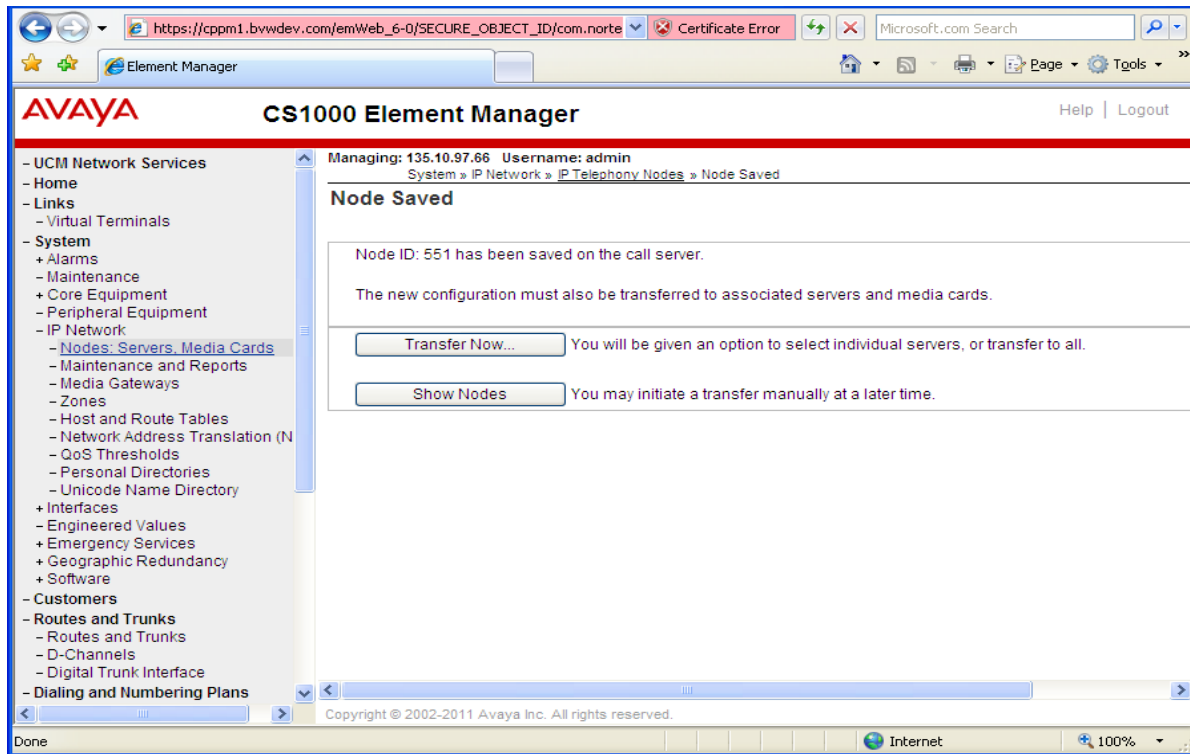
**Figure 21: Node ID: 551 – Virtual Trunk Gateway Configuration Details**

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**.



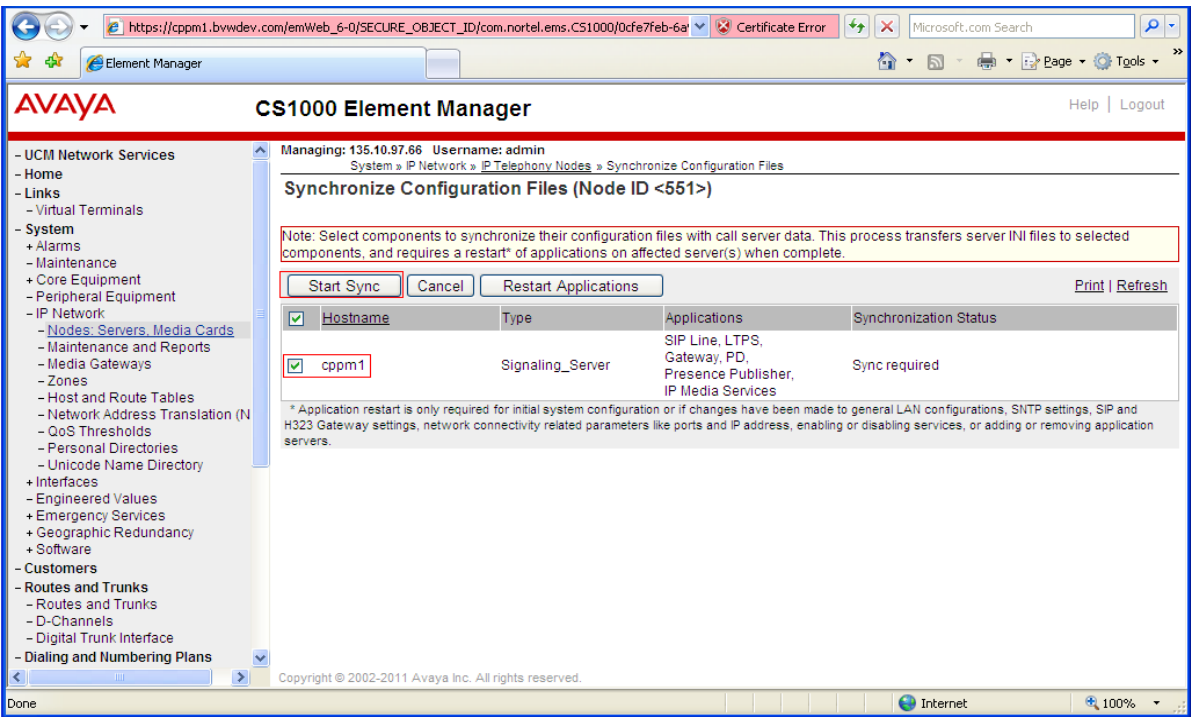
**Figure 22: The SIP Gateway Setting of Telephony Node**

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 **Add** button.

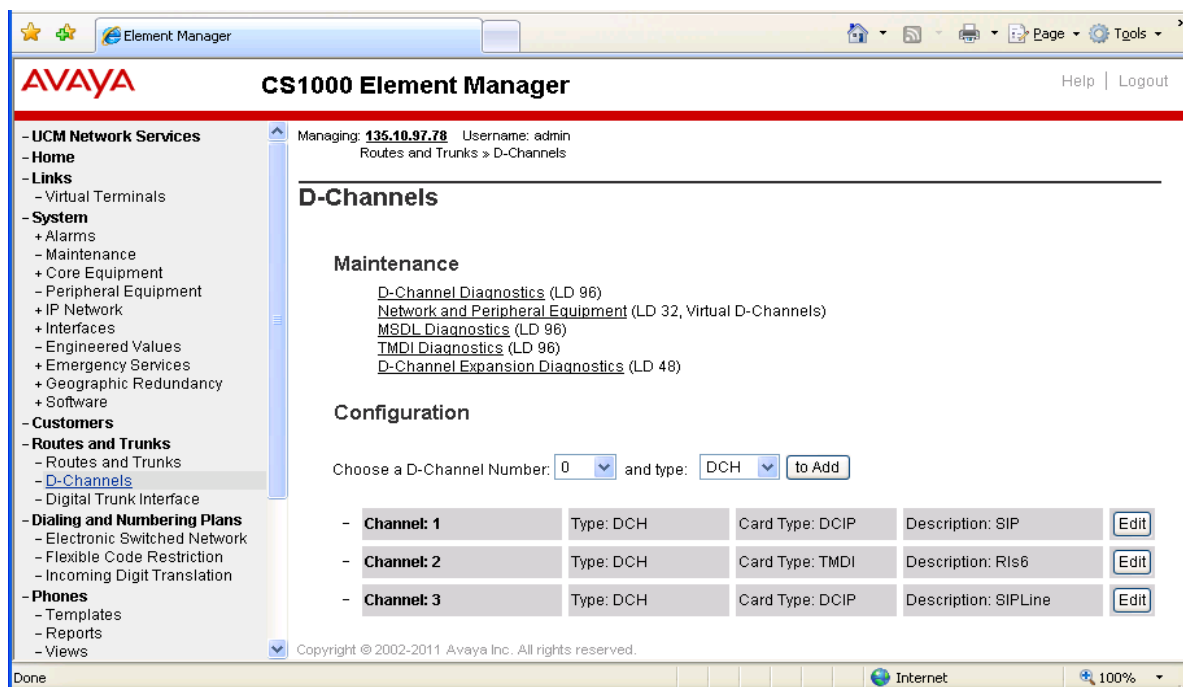


Figure 25: D-Channels page

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.

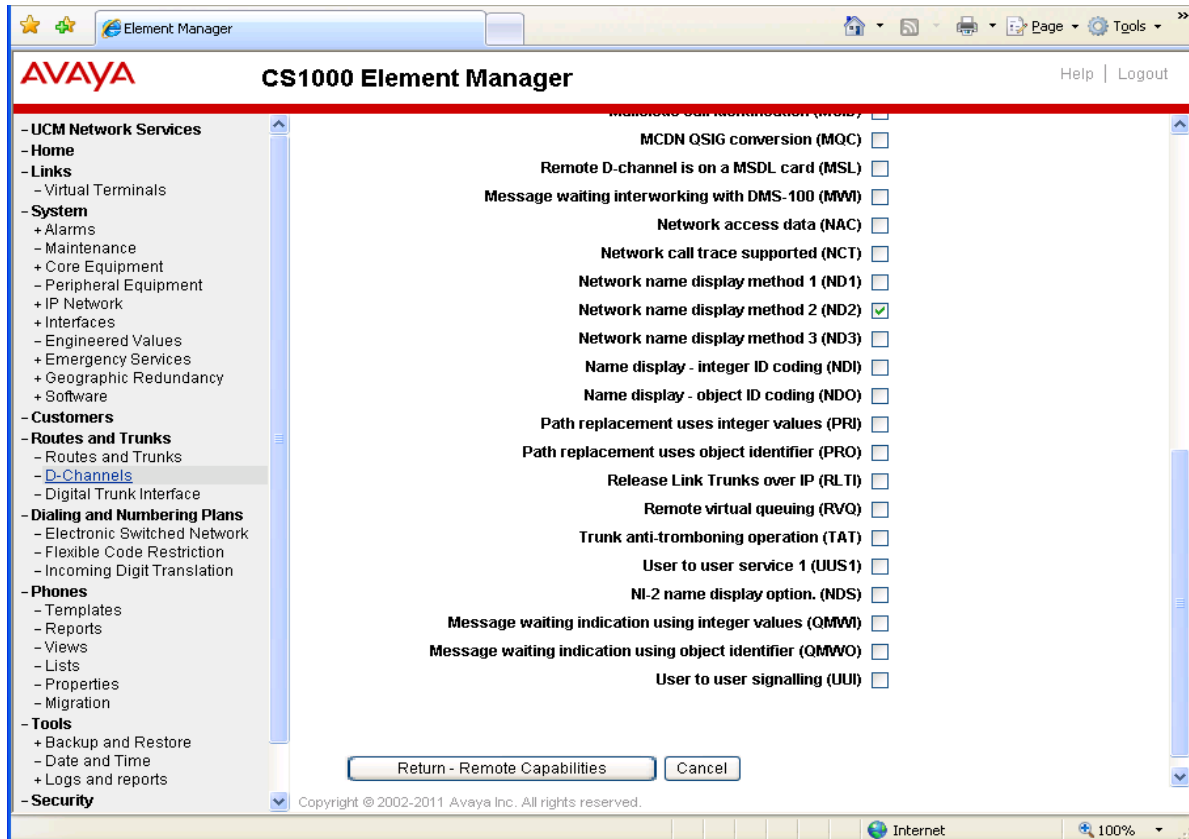
Input Description	Input Value
Action Device And Number (ADAN):	DCH
D channel Card Type:	D-Channel is over IP (DCIP) *
Designator:	SIP
Recovery to Primary:	<input type="checkbox"/>
PRI loop number for Backup D-channel:	
User:	Integrated Services Signaling Link Dedicated (ISLD) *
Interface type for D-channel:	Meridian DMS-100 (D100)
Country:	ETS 300 =102 basic protocol (ETSI)
D-Channel PRI loop number:	
Primary Rate Interface:	<input type="text"/> <a href="#">more PRI</a>
Secondary PRI2 loops:	<input type="text"/>
Release ID of the switch at the far end:	25
Central Office switch type:	100% compatible with Bellcore standard (STD)
Integrated Services Signaling Link Maximum:	4000 Range: 1 - 4000
Signalling server resource capacity:	3700 Range: 0 - 3700

+ Basic options (BSCOPT)  
+ Advanced options (ADVOPT)  
+ Feature Packages

**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**

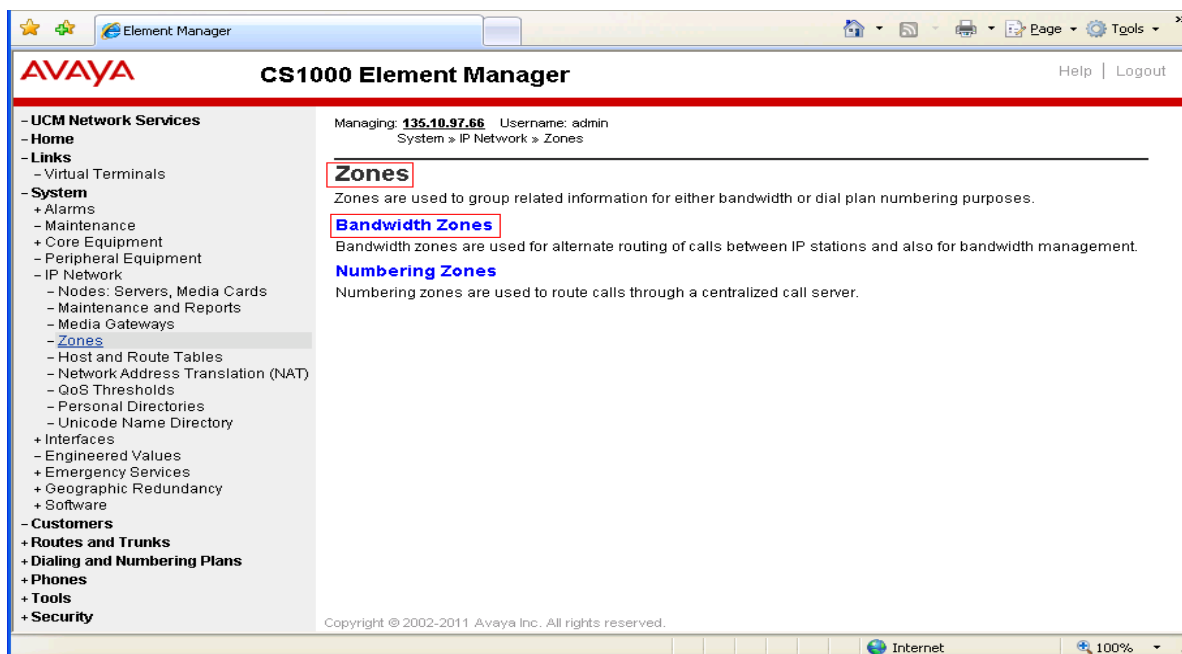
Continue expanding the **Advanced options (ADVOPT)** subsection of the **Basic Configuration** section, the **Advanced options** section appears as shown in **Figure 28** below. Leave all the values at default.

**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.

AVAYA CS1000 Element Manager

Managing: 135.10.97.66 Username: admin  
System » IP Network » Zones » Bandwidth Zones » Zone Basic Property and Bandwidth Management

### Zone Basic Property and Bandwidth Management

Input Description	Input Value
Zone Number (ZONE):	4 * ( 1 - 8000 )
Intrazone Bandwidth (INTRA_BW):	1000000 ( 0 - 10000000 )
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	1000000 ( 0 - 10000000 )
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	VTRK (VTRK)
Description (ZDES):	For_Virtual_Trunks

\* Required value.

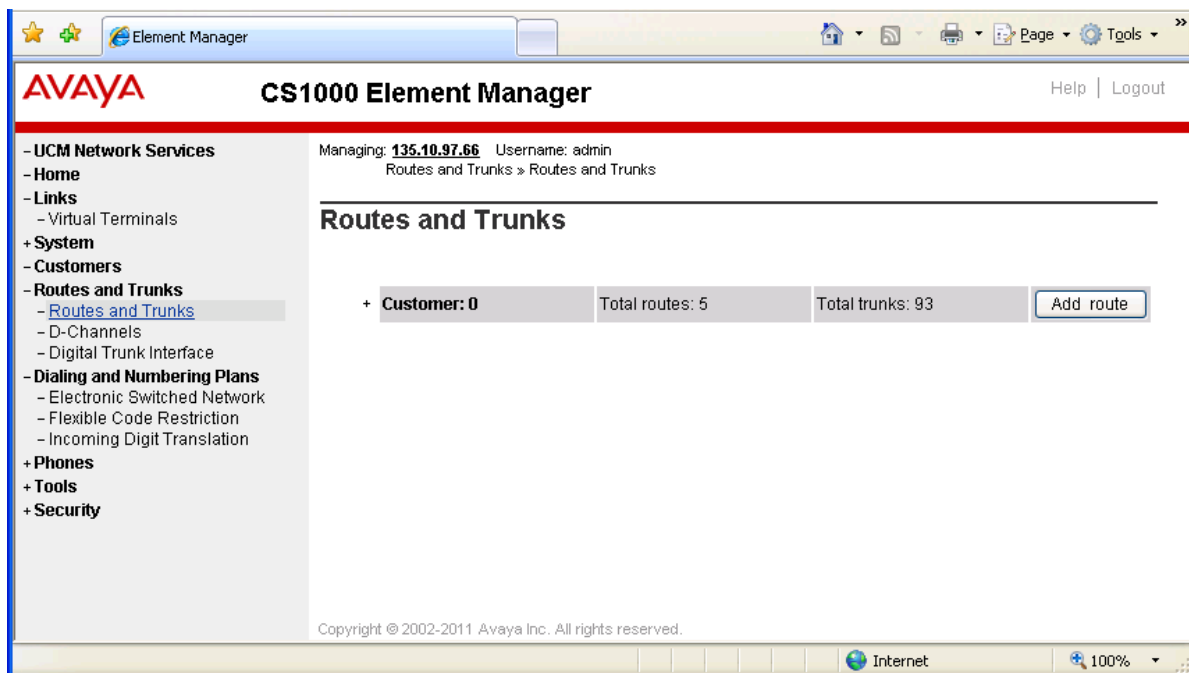
Save Cancel

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**Figure 30: Zone Basic Property and Bandwidth Management**

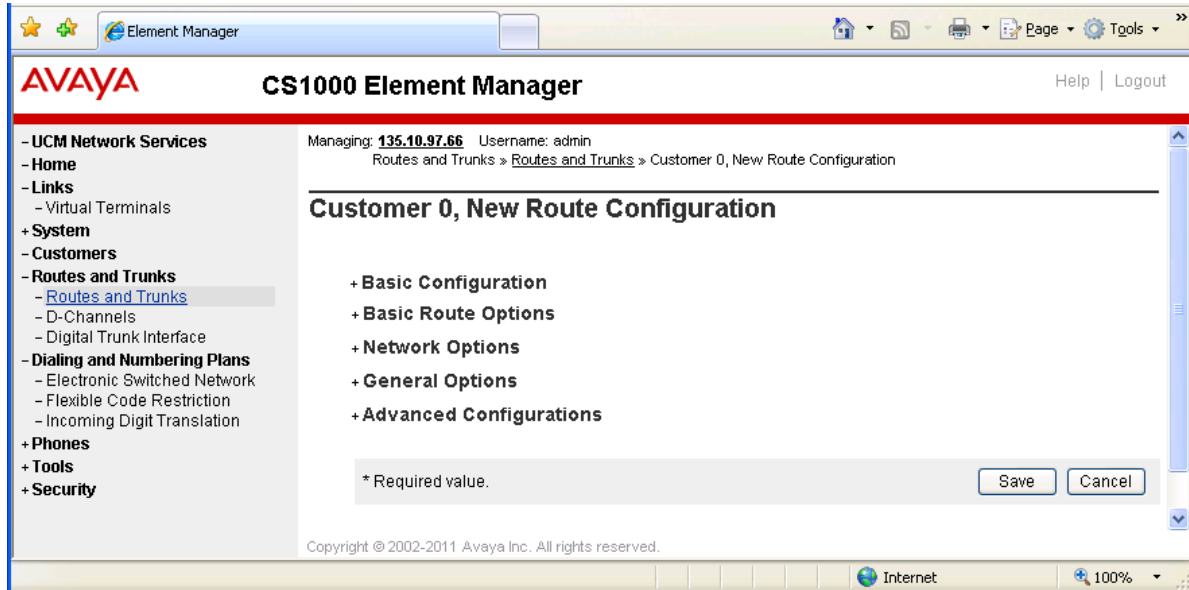
## 6.4. Configure SIP Route

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 is 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.

The screenshot shows the Avaya CS1000 Element Manager interface. The left sidebar contains a navigation tree with categories like UCM Network Services, System, Customers, Routes and Trunks, and Dialing and Numbering Plans. The main content area is titled 'Basic Configuration' and contains the following fields and options:

- Route data block (RDB) (TYPE): RDB
- Customer number (CUST): 0
- Route number (ROUT): 1
- Designator field for trunk (DES): SIP
- Trunk type (TKTP): TIE trunk data block (TIE)
- Incoming and outgoing trunk (ICOG): Incoming and Outgoing (IAO)
- Access code for the trunk route (ACOD): 8001
- Trunk type M911P (M911P): ☐
- The route is for a virtual trunk route (VTRK): ☒
- Zone for codec selection and bandwidth management (ZONE): 4 (0 - 8000)
- Node ID of signaling server of this route (NODE): 551 (0 - 9999)
- Protocol ID for the route (PCID): SIP (SIP)
- Print correlation ID in CDR for the route (CRID): ☐
- Integrated services digital network option (ISDN): ☐
- Calling number dialing plan (CNDP): Coordinated dialing plan (CDP)

Below the configuration fields, there are expandable sections for '+ Basic Route Options', '+ Network Options', '+ General Options', and '+ Advanced Configurations'. The footer of the page includes the copyright notice 'Copyright © 2002-2011 Avaya Inc. All rights reserved.' and a status bar showing 'Internet' and '100%' zoom.

**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.

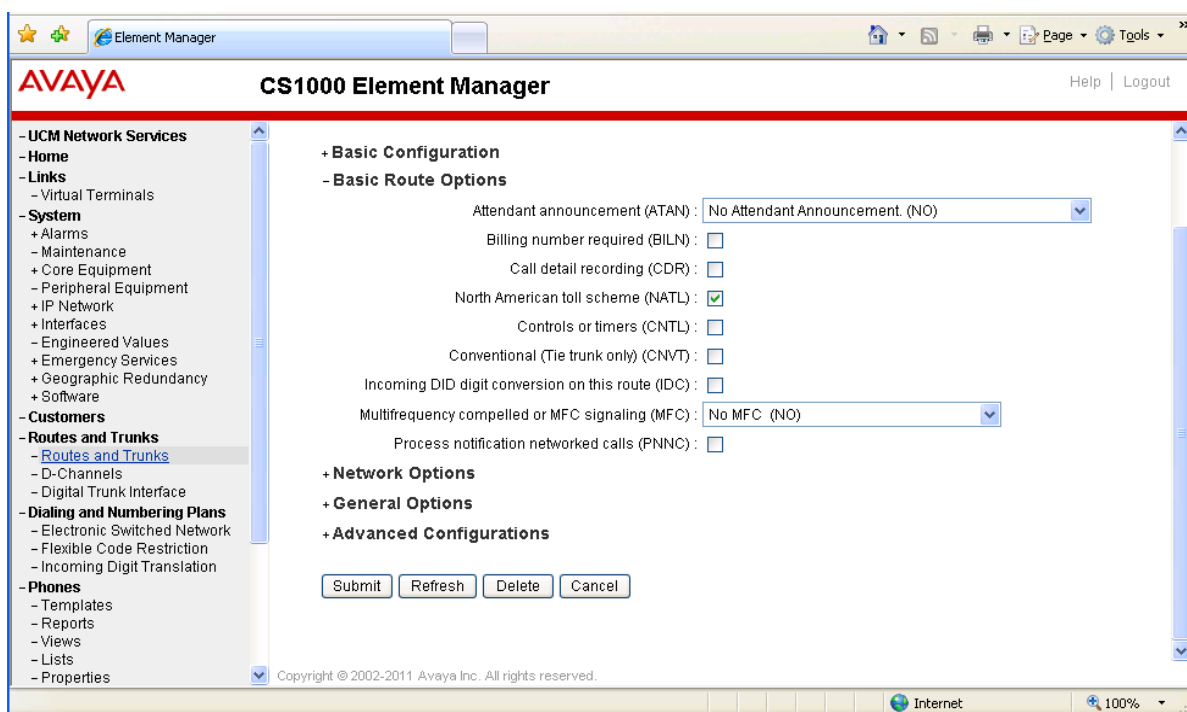
- **Mode of operation (MODE):** *Route uses ISDN Signaling Link (ISLD)*
- **D channel number (DCH):** *10* - this is D-Channel for SIP Trunk as defined in the **Section 6.2**
- **Interface type for route (IFC):** *Meridian 1 (SL1)*
- **Private Network Identifier (PNI):** *1*
- **Network calling name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

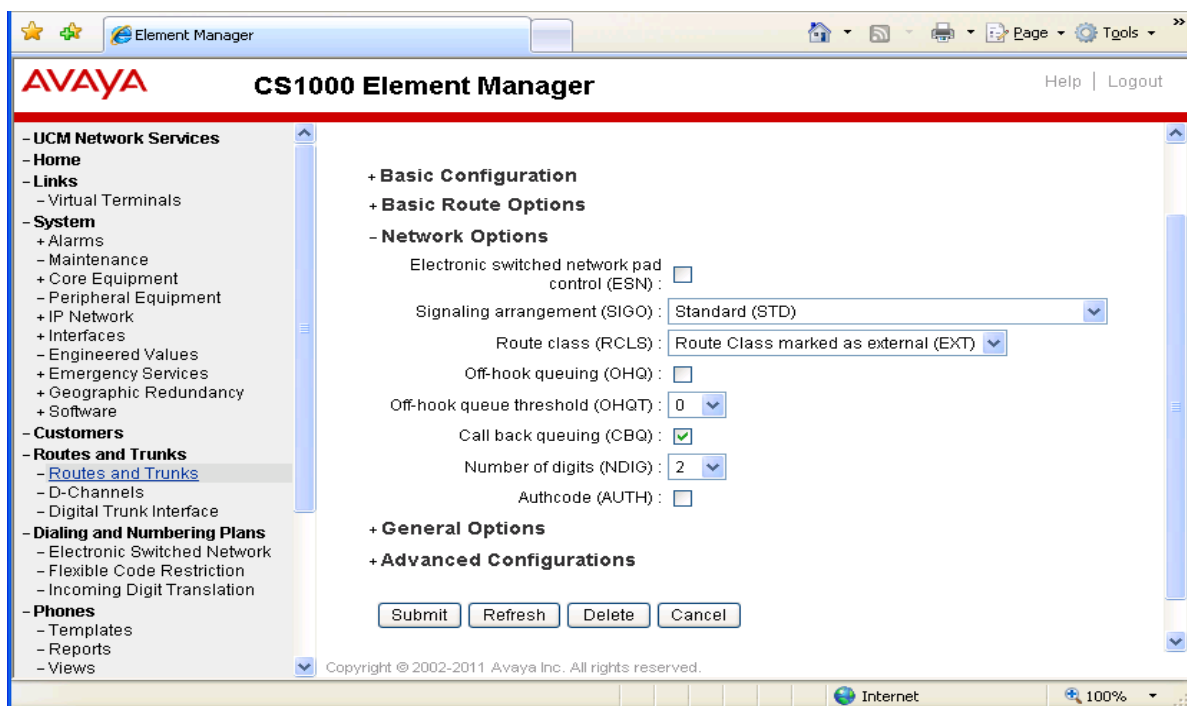


Continue expanding the **Basic Route Options** section and keep default values as shown in **Figure 35**.



**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.

The screenshot displays the AVAYA CS1000 Element Manager web interface. The left sidebar contains a navigation tree with the following sections: UCM Network Services, Home, Links, System, Customers, Routes and Trunks (highlighted), Dialing and Numbering Plans, and Phones. The main content area is titled '+ Network Options' and contains the '- General Options' section. This section includes several configuration items: 'M1 is the only controlling party on incoming calls (CPDC)' (checkbox), 'Dial tone on originating calls (DLTN)' (checkbox), 'Hold failure threshold (HOLD)' (text field with value '02 02 40'), 'Trunk access restriction group (TARG)' (text field with value '01'), 'Alternate trunk route for outgoing trunks (STEP)' (text field with value '(0 - 511)'), 'Actual outgoing toll digits to be ignored for code restriction (OABS)' (text field), 'Display IDC name (DNAM)' (checkbox), 'Enable equal access restrictions (EQAR)' (checkbox), 'ACD DNIS route (DNIS)' (checkbox), and 'Include DNIS number in CDR records (DCDR)' (checkbox). Below these options is the '+ Advanced Configurations' section, which is currently collapsed. At the bottom of the main content area are four buttons: 'Submit', 'Refresh', 'Delete', and 'Cancel'. The footer of the interface shows the copyright notice 'Copyright © 2002-2011 Avaya Inc. All rights reserved.' and the status bar indicates 'Internet' and '100%' zoom.

AVAYA CS1000 Element Manager

Help | Logout

- UCM Network Services

- Home

- Links

- Virtual Terminals

- System

+ Alarms

- Maintenance

+ Core Equipment

- Peripheral Equipment

+ IP Network

+ Interfaces

- Engineered Values

+ Emergency Services

+ Geographic Redundancy

+ Software

- Customers

- Routes and Trunks

- Routes and Trunks

- D-Channels

- Digital Trunk Interface

- Dialing and Numbering Plans

- Electronic Switched Network

- Flexible Code Restriction

- Incoming Digit Translation

- Phones

- Templates

- Reports

- Views

+ Network Options

- General Options

M1 is the only controlling party on incoming calls (CPDC) : ☐

Dial tone on originating calls (DLTN) : ☐

Hold failure threshold (HOLD) : 02 02 40

Trunk access restriction group (TARG) : 01

Alternate trunk route for outgoing trunks (STEP) : (0 - 511)

Actual outgoing toll digits to be ignored for code restriction (OABS) :

Display IDC name (DNAM) : ☐

Enable equal access restrictions (EQAR) : ☐

ACD DNIS route (DNIS) : ☐

Include DNIS number in CDR records (DCDR) : ☐

+ Advanced Configurations

Submit Refresh Delete Cancel

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Internet 100%

**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**.

**AVAYA CS1000 Element Manager**

Help | Logout

- UCM Network Services
- Home
- Links
  - Virtual Terminals
- System
  - + Alarms
  - Maintenance
  - + Core Equipment
  - Peripheral Equipment
  - + IP Network
  - + Interfaces
  - Engineered Values
  - + Emergency Services
  - + Geographic Redundancy
  - + Software
- Customers
- Routes and Trunks
  - **Routes and Trunks**
  - D-Channels
  - Digital Trunk Interface
- Dialing and Numbering Plans
  - Electronic Switched Network
  - Flexible Code Restriction
  - Incoming Digit Translation
- Phones
  - Templates
  - Reports
  - Views
  - Lists
  - Properties
  - Migration
- Tools
  - + Backup and Restore

**- Advanced Configurations**

Malicious call trace alarm is allowed for ☐ external calls (ALRM) :

Allow last re-directing number (ARDN) :

ANI identifier number (ANTK) :

AC15 timed reminder recall (ATTR) : ☐

Auto terminate (AUTO) : ☐

Collect call blocking allowed (CCBA) : ☐

Call forward restriction (CFWR) : ☐

Maximum number of CNI digits (CLEN) :

Time (in seconds) that an extension is allowed to ring or be On-hold or Call Park before the trunk is disconnected (DCT) :  (0 - 511)

North American distinctive ringing for incoming calls (DRNG) : ☐

Home local number (HLCL) :

Home national number (HNTN) :

In-band automatic number identification route ☐

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Internet 100%

**Figure 38: Advanced Configurations of Route**

**AVAYA CS1000 Element Manager**

Help | Logout

- UCM Network Services
- Home
- Links
  - Virtual Terminals
- System
  - + Alarms
  - Maintenance
  - + Core Equipment
  - Peripheral Equipment
  - + IP Network
  - + Interfaces
  - Engineered Values
  - + Emergency Services
  - + Geographic Redundancy
  - + Software
- Customers
- Routes and Trunks
  - **Routes and Trunks**
  - D-Channels
  - Digital Trunk Interface
- Dialing and Numbering Plans
  - Electronic Switched Network
  - Flexible Code Restriction
  - Incoming Digit Translation
- Phones
  - Templates
  - Reports
  - Views
  - Lists
  - Properties
  - Migration
- Tools
  - + Backup and Restore
  - Date and Time

**- Advanced Configurations**

In-band automatic number identification route (IANI) : ☐

Incoming identifier send (ICIS) : ☒

Internal/external definition (IDEF) :

Identify originating party (IDOP) : ☐

Insert (INST) :

Manual outgoing trunk route (MANO) : ☐

Manual route (MNL) : ☐

Music on-hold (MUS) : ☐

Outgoing identifier send (OGIS) : ☒

Off-hook timer delay (OHTD) : ☐

Outpulsing route (OPR) : ☐

Pseudo answer (PANS) : ☒

Periodic clearing signal (PECL) : ☐

Privacy indicator ignored (PII) : ☐

Auxiliary application (AUXP) : ☐

Priority level (PLEV) :

Protocol selection (PSEL) :

Preference trunk usage threshold (PTUT) :  (0 - 510)

Port type at far end (PTYT) :

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Internet 100%

**Figure 39: Advanced Configurations of new Route (cont)**

**AVAYA CS1000 Element Manager** Help | Logout

- UCM Network Services
  - Home
  - Links
    - Virtual Terminals
  - System
    - + Alarms
    - Maintenance
    - + Core Equipment
    - Peripheral Equipment
    - + IP Network
    - + Interfaces
    - Engineered Values
    - + Emergency Services
    - + Geographic Redundancy
    - + Software
  - Customers
  - Routes and Trunks
    - [Routes and Trunks](#)
    - D-Channels
    - Digital Trunk Interface
  - Dialing and Numbering Plans
    - Electronic Switched Network
    - Flexible Code Restriction
    - Incoming Digit Translation
  - Phones
    - Templates
    - Reports
    - Views
    - Lists
    - Properties
    - Migration
  - Tools
    - + Backup and Restore
    - Date and Time

Port type at far end (PTYP) : Analog TIE trunks (ATT)

Route traffic information in ACD Reports (RACD) : ☐

Radio paging route (RPA) : ☐

Route number (RTN) :  (0 - 511)

Satellite used for trunk route (SAT) : ☐

Scheduled access restriction group (SGRP) :  (0 - 999)

Special service list number (SSL) :

Standard signaling type (STYP) : Standard Data (SDAT)

CPP/CPPO flag for incoming non-ISDN trunk call tandemed to this trunk route (TCPP) : ☐

Tone detector required (TDET) : ☐

Trunk identity (TIDY) :  8000 1

Tromboning (TRMB) : ☒

Recall signal (may not) may be received and transmitted on this route (TRRL) : ☐

Tone table number (TTBL) :  0

Answer an attendant extended call over VNS immediately on the incoming bearer trunk (VRAT) : ☐

Incoming CLID Table (CTBL) :  (0 - 256)

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**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.

The screenshot displays the Avaya CS1000 Element Manager web interface. The top navigation bar includes the Avaya logo, the title 'CS1000 Element Manager', and links for 'Help' and 'Logout'. Below this, a breadcrumb trail shows the path: 'Managing: 135.10.97.78 Username: admin' > 'Routes and Trunks > Routes and Trunks' > 'Customer 0, Route 1'. The main content area is titled 'Customer 0, Route 1, Trunk type TIE trunk data block'. On the left, a sidebar menu lists various configuration categories: UCM Network Services, Home, Links, System, Customers, Routes and Trunks (selected), Dialing and Numbering Plans, Phones, and Tools. The 'Routes and Trunks' section is expanded, showing 'Routes and Trunks' as the active sub-menu. The main configuration area is divided into two sections: 'Basic Configuration' and '+Advanced Trunk Configurations'. The 'Basic Configuration' section contains the following fields: 'Multiple trunk input number' (32), 'Range: 2 - 3700', 'Auto increment member number' (checked), 'Trunk data block' (IP Trunk (IPT)), 'Terminal number' (100 0 0 0), 'Designator field for trunk' (SIP), 'Extended trunk' (VTRK), 'Member number' (1), 'Level 3 Signaling' (dropdown), 'Card density' (Octal Density (8D)), 'Start arrangement Incoming' (Immediate (IMM)), 'Start arrangement Outgoing' (Immediate (IMM)), 'Trunk group access restriction' (1), and 'Channel ID for this trunk' (1). The 'Class of Service' field has an 'Edit' button. The '+Advanced Trunk Configurations' section is currently collapsed. The footer of the interface shows the copyright notice: 'Copyright © 2002-2011 Avaya Inc. All rights reserved.'

**Figure 41: Basic Configuration of new Trunk**

**AVAYA CS1000 Element Manager** Help | Logout

- UCM Network Services
  - Home
  - Links
    - Virtual Terminals
  - System
    - + Alarms
    - + Maintenance
    - + Core Equipment
    - + Peripheral Equipment
    - + IP Network
    - + Interfaces
    - + Engineered Values
    - + Emergency Services
    - + Geographic Redundancy
    - + Software
  - Customers
  - Routes and Trunks
    - Routes and Trunks
    - D-Channels
    - Digital Trunk Interface
  - Dialing and Numbering Plans
    - Electronic Switched Network
    - Flexible Code Restriction
    - Incoming Digit Translation
  - Phones
    - Templates
    - Reports
    - Views

Input Description	Input Value
- ACD Priority :	ACD Priority not required (APN) ▾
- Analog Semi-Permanent Connections :	Analog Semi-Permanent Connections Denied (SPCD) ▾
- ARF Supervised COT:	▾
- Barring:	▾
- Battery Supervised COT :	▾
- Busy Tone Supervised COT:	▾
- Calling Line Identification:	Calling Line Identification Allowed (CLIA) ▾
- Calling party:	Calling party Allowed (CNA) ▾
- Central Office Ringback:	▾
- Centrex Switchhook Flash:	Centrex Switchhook Flash Denied (THFD) ▾
- Dial Pulse:	Dial Pulse (DIP) ▾
- DTR PAD value:	▾
- Echo Canceling:	Echo Canceling Denied (ECD) ▾
- Hong Kong DTI :	▾

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Done Internet 100%

**Figure 42: Class of Service of new Trunk**

**AVAYA CS1000 Element Manager** Help | Logout

- UCM Network Services
  - Home
  - Links
    - Virtual Terminals
  - System
    - + Alarms
    - + Maintenance
    - + Core Equipment
    - + Peripheral Equipment
    - + IP Network
    - + Interfaces
    - + Engineered Values
    - + Emergency Services
    - + Geographic Redundancy
    - + Software
  - Customers
  - Routes and Trunks
    - Routes and Trunks
    - D-Channels
    - Digital Trunk Interface
  - Dialing and Numbering Plans
    - Electronic Switched Network
    - Flexible Code Restriction
    - Incoming Digit Translation
  - Phones
    - Templates
    - Reports
    - Views
    - Lists
    - Properties
    - Migration
  - Tools

- Hong Kong DTI :	▾
- Loop Break Supervised COT:	▾
- Make-break ratio for dial pulse:	10 pulses per second (P10) ▾
- Manual Incoming:	Manual Incoming Denied (MID) ▾
- Media Security:	Media Security Never (MSNV) ▾
- Network Hook Flash Over M911P:	▾
- Polarity:	▾
- Priority:	Low Priority (LPR) ▾
- Restriction level:	Unrestricted (UNR) ▾
- Reversed Ear Piece:	Reversed Ear Piece denied (XREP) ▾
- Short or long line:	▾
- Transmission Class of Service:	Non-Transmission Compensated (NTC) ▾
- Warning Tone:	Warning Tone Allowed (WTA) ▾
- Reversed Ear Piece:	Reversed Ear Piece denied (XREP) ▾
- ARF Supervised COT:	▾

Return Class of Service Cancel

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Done Internet 100%

**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.

**AVAYA CS1000 Element Manager** Help | Logout

**- UCM Network Services**

- Home
- Links
  - Virtual Terminals
- **System**
  - + Alarms
  - Maintenance
  - + Core Equipment
  - Peripheral Equipment
  - + IP Network
  - + Interfaces
  - Engineered Values
  - + Emergency Services
  - Geographic Redundancy
  - + Software
- **Customers**
- **Routes and Trunks**
  - Routes and Trunks
  - D-Channels
  - Digital Trunk Interface
- **Dialing and Numbering Plans**
  - Electronic Switched Network
  - Flexible Code Restriction
  - Incoming Digit Translation
- **Phones**
  - Templates
  - Reports
  - Views
  - Lists
  - Properties
  - Migration
- **Tools**
  - + Backup and Restore
  - Date and Time
  - + Logs and reports

**- Advanced Trunk Configurations**

CTI trunk Monitoring and Control: ☐

Auto Terminate DN:

Music conference loop:  ( 0 - 159 )

Call modification features restriction: ☐

Digit collection ready: ☐

Forced Charge Account: ☐

Multifrequency digit level: 0

Multifrequency PAD: ☐

Manual Directory Number:

Network Class of Service group: 7

Night service group number: 0

Night service directory number:

Pulse code modulation law:

Pad category table number for digital trunks: 1

Private line directory number:

Is the ISPC link used by a D-channel: ☐

Signaling category table number: 1

Connection Reference Number:  ( 1 - 9999999 )

Answer and disconnect supervision required: ☒

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**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.

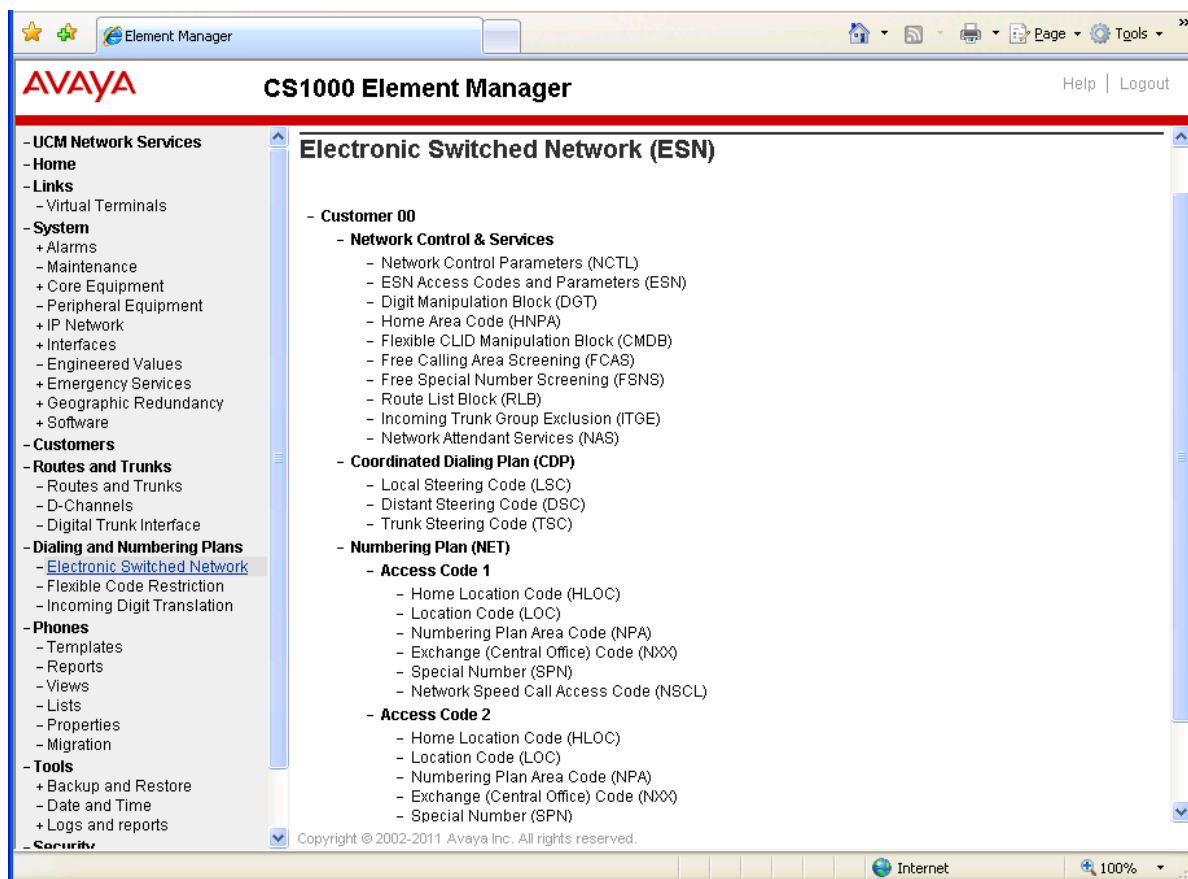
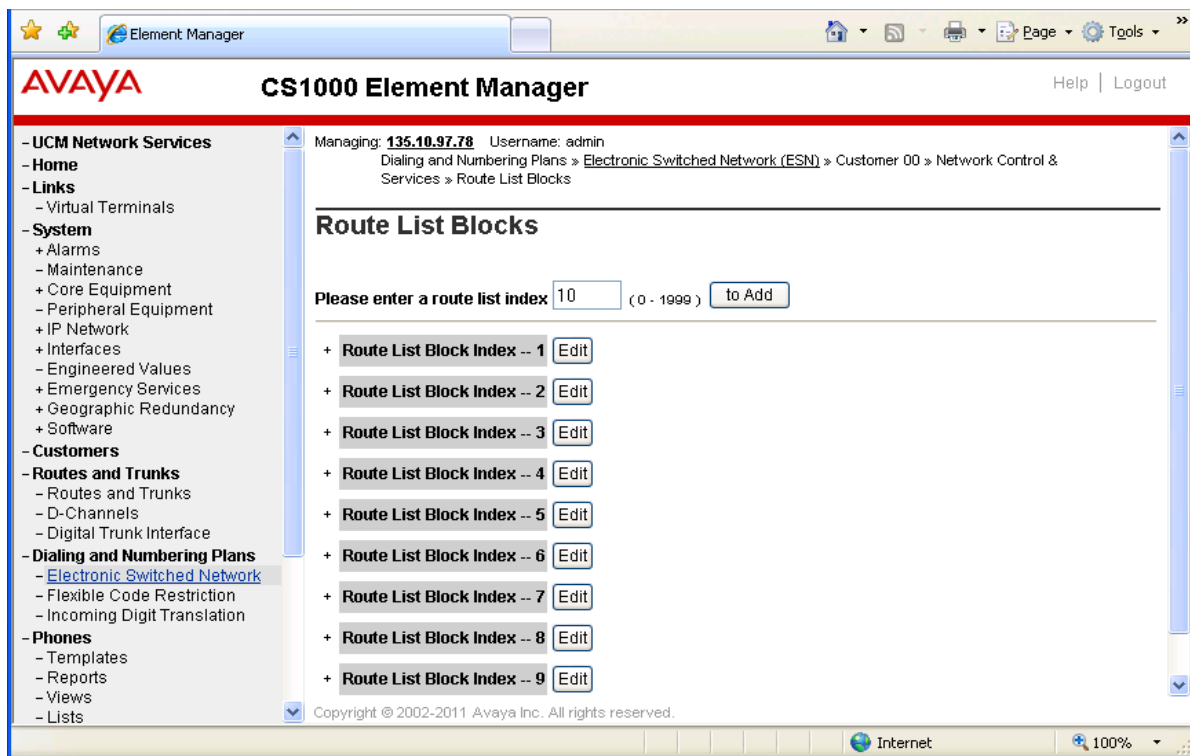


Figure 45: Electronic Switched Network (ESN) page



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.

**AVAYA CS1000 Element Manager**

Help | Logout

- UCM Network Services
  - Home
  - Links
    - Virtual Terminals
  - System
    - + Alarms
    - Maintenance
    - + Core Equipment
    - Peripheral Equipment
    - + IP Network
    - + Interfaces
    - Engineered Values
    - + Emergency Services
    - + Geographic Redundancy
    - + Software
  - Customers
    - Routes and Trunks
      - Routes and Trunks
      - D-Channels
      - Digital Trunk Interface
  - Dialing and Numbering Plans
    - **Electronic Switched Network**
    - Flexible Code Restriction
    - Incoming Digit Translation
  - Phones
    - Templates
    - Reports
    - Views
    - Lists
    - Properties
    - Migration
  - Tools
    - + Backup and Restore
    - Date and Time
    - + Logs and reports
  - Security

**Route List Block**

**General Properties**

Number of Alternate Routing Attempts: 5 (1 - 10)

Initial Set: 0 (0 - 64)

Set Minimum Facility Restriction Level:

Overlap Length: 0 (0 - 24)

Extended Local Calls: ☐

Route List Index: 10

Entry Number for the Route List: 0 (0 - 63)

**Indexes**

Time of Day Schedule: 0

Facility Restriction Level: 0 (0 - 7)

Digit Manipulation Index: 0

ISL D-Channel Down Digit Manipulation Index: 0 (0 - 1999)

Free Calling Area Screening Index: 0

Free Special Number Screening Index: 0

Business Network Extension Route: ☐

Incoming CLID Table: 0 (0 - 100)

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Internet 100%

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

In the **Options** and **VNS 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.

The screenshot displays the AVAYA CS1000 Element Manager web interface. The left sidebar contains a navigation menu with categories like UCM Network Services, Links, System, Customers, Routes and Trunks, Dialing and Numbering Plans, Phones, and Tools. The main content area is titled 'Options' and contains the following settings:

- Local Termination entry: ☐
- Route Number: 1 (dropdown menu)
- Skip Conventional Signaling: ☐
- Display Originator's Information: ☐
- Use Tone Detector: ☐
- Conversion to LDN: ☐
- Expensive Route: ☐
- Strategy on Congestion: No Reroute (NRR) (dropdown menu)
- QSIG Alternate Routing Causes: QSIG Alternate Routing Cause 1 (dropdown menu)
- Preferred Routing: Preferred Route 1 (dropdown menu)
- ISDN Drop Back Busy: Drop Back Disabled (DBD) (dropdown menu)
- ISDN Off-Hook Queuing Option: ☐
- Off-Hook Queuing Allowed: ☐
- Call Back Queuing Allowed: ☐

Below the 'Options' section is the 'VNS Options' section with the following setting:

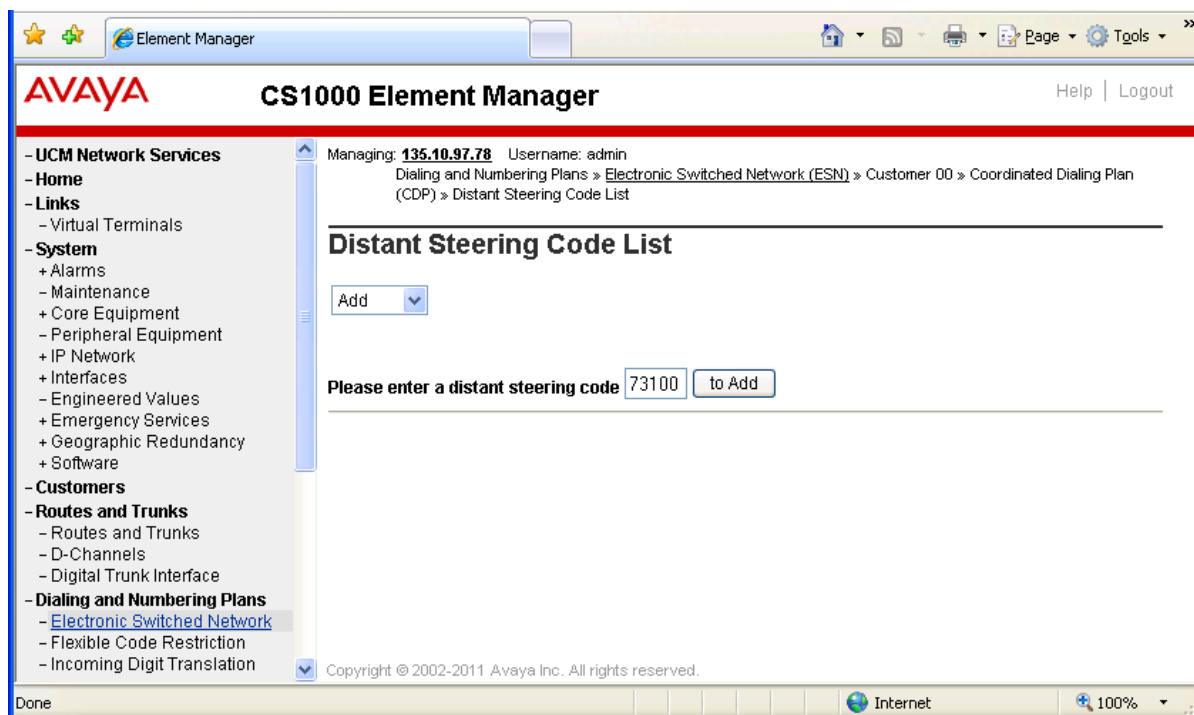
- Entry is a VNS Route: ☐

At the bottom right of the main content area are 'Submit' and 'Cancel' buttons. The footer of the page includes the copyright notice 'Copyright © 2002-2011 Avaya Inc. All rights reserved.' and a status bar showing 'Internet' and '100%' zoom.

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

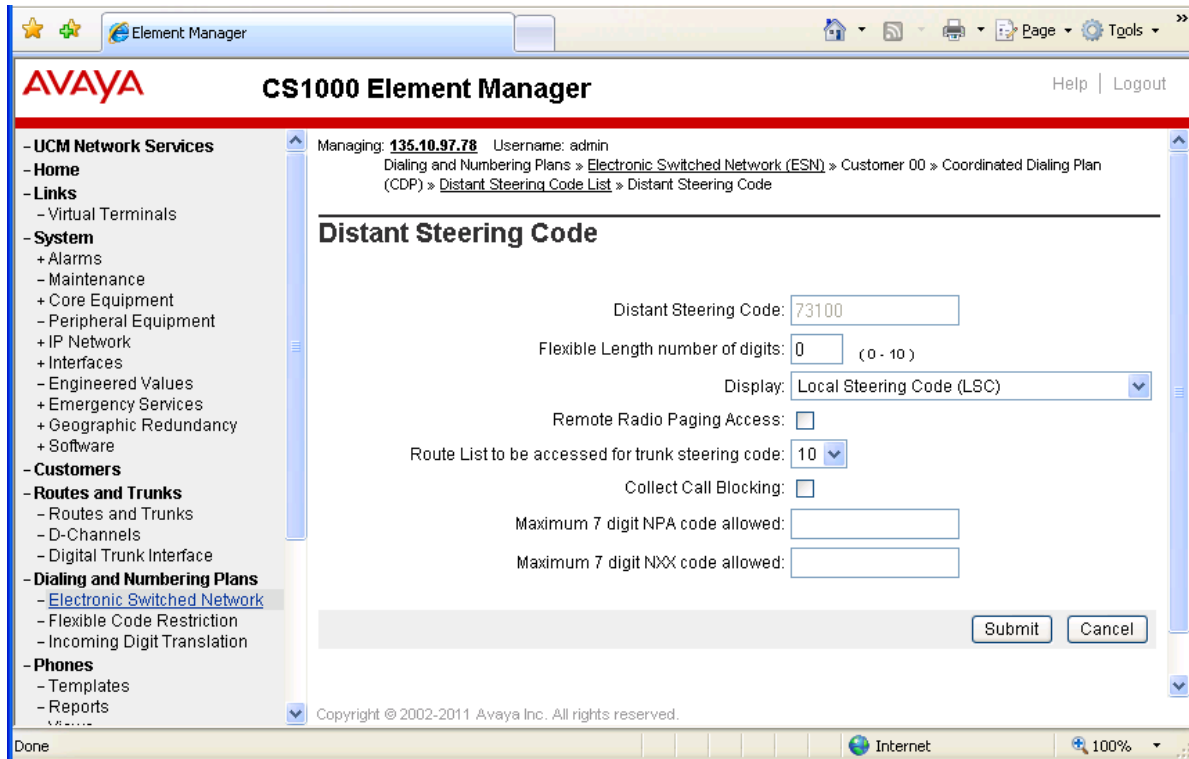
## 6.6.2. Create a Distant Steering Code (DSC)

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.



AVAYA CS1000 Element Manager

Managing: 135.10.97.78 Username: admin  
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Coordinated Dialing Plan (CDP) » Distant Steering Code List » Distant Steering Code

### Distant Steering Code

Distant Steering Code: 73100

Flexible Length number of digits: 0 (0 - 10)

Display: Local Steering Code (LSC)

Remote Radio Paging Access: ☐

Route List to be accessed for trunk steering code: 10

Collect Call Blocking: ☐

Maximum 7 digit NPA code allowed:

Maximum 7 digit NXX code allowed:

Submit Cancel

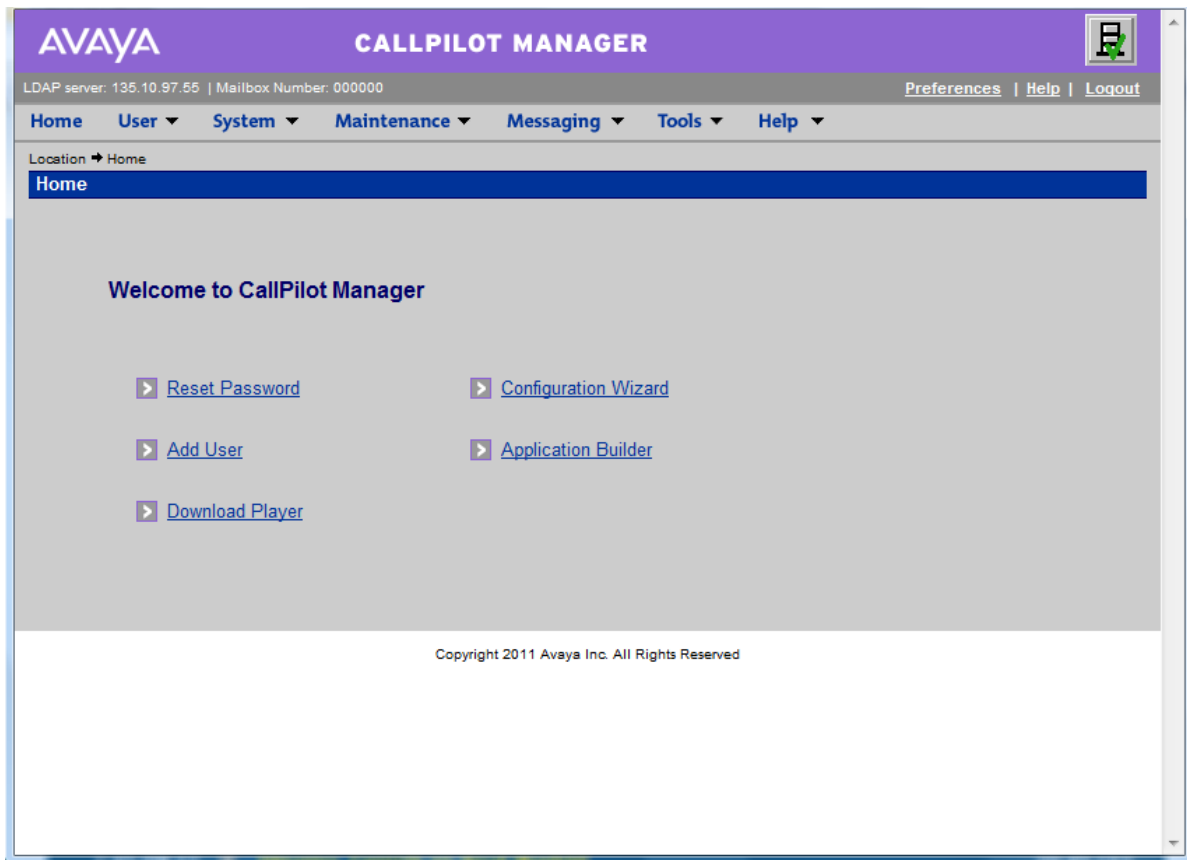
Copyright © 2002-2011 Avaya Inc. All rights reserved.

**Figure 50: Distant Steering Code page**

## 7. Configure Avaya CallPilot® Through-Dial

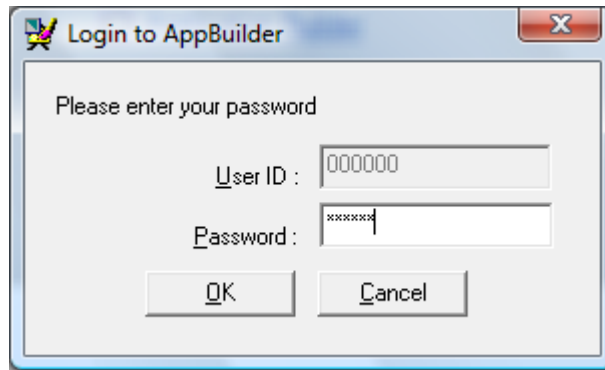
This document assumes that the Avaya CallPilot messaging system was properly installed, configured and administered as per the product document, for more information about how to install, configure and administer the CallPilot system please refer to **Section 11 [8]**. This section provides the steps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



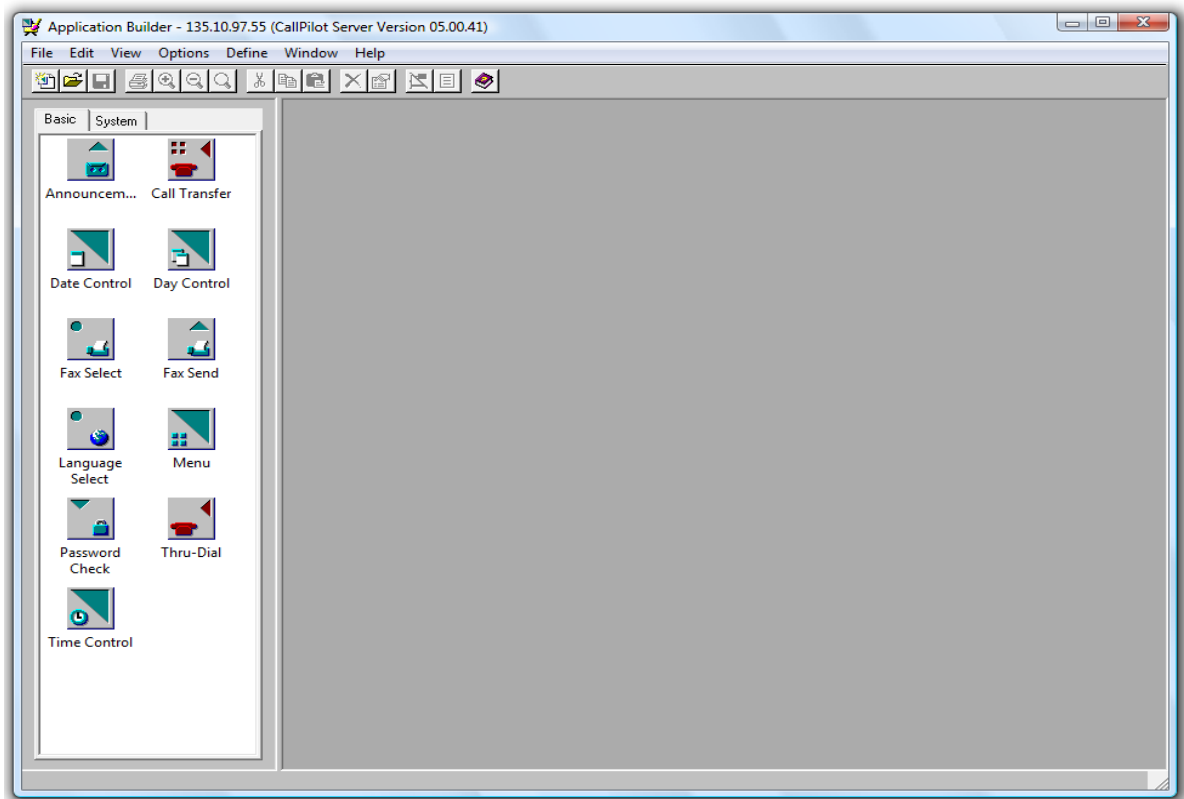
**Figure 51: CallPilot Manager Home page**

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.



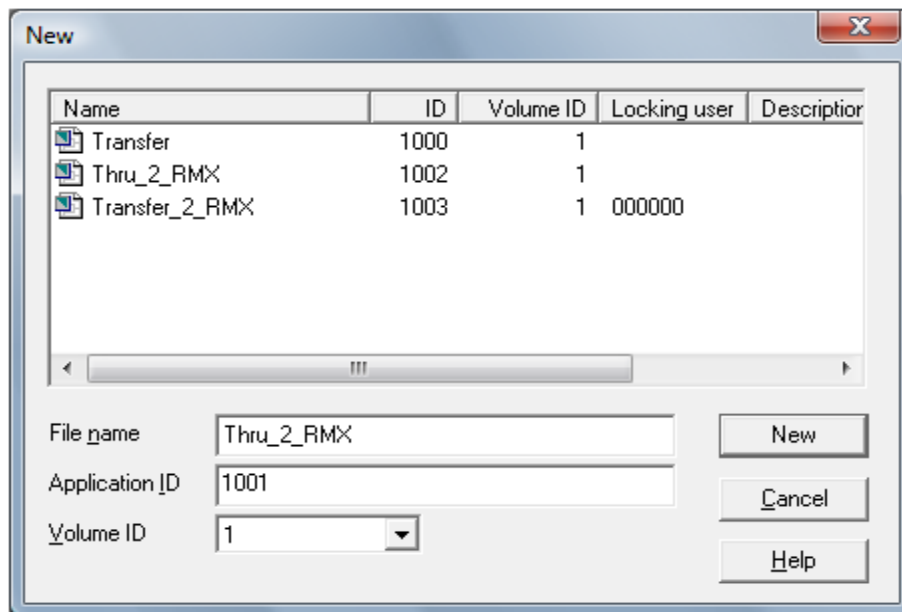
**Figure 52: Login window of Application Builder**

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



**Figure 53: Application Builder window**

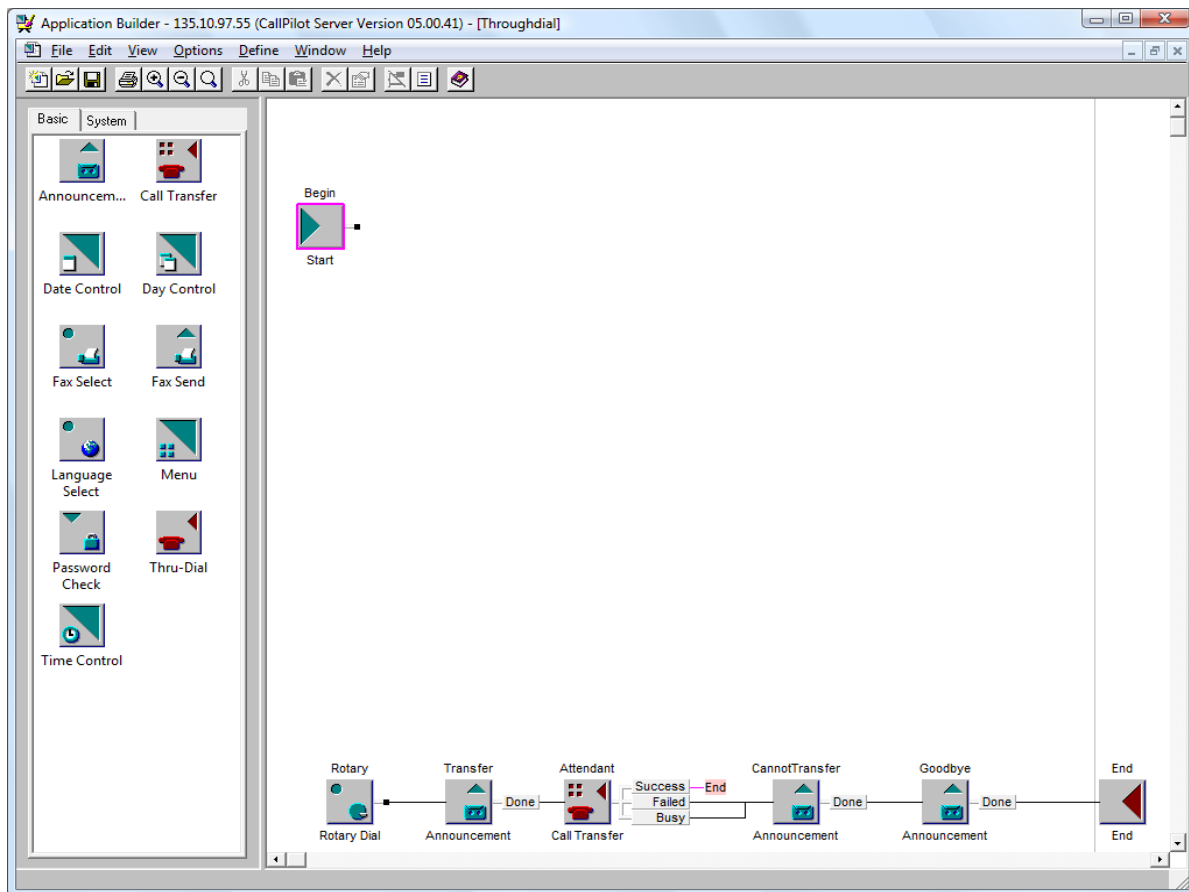
From the **Application Builder** window, navigate to menu **File > New** to create a new application; the **New** application window appears as shown in **Figure 54**. Enter a name in the **File name** box, for example *Thru\_2\_RMX*, and then click on the **New** button.



**Figure 54: New window of Application Builder**

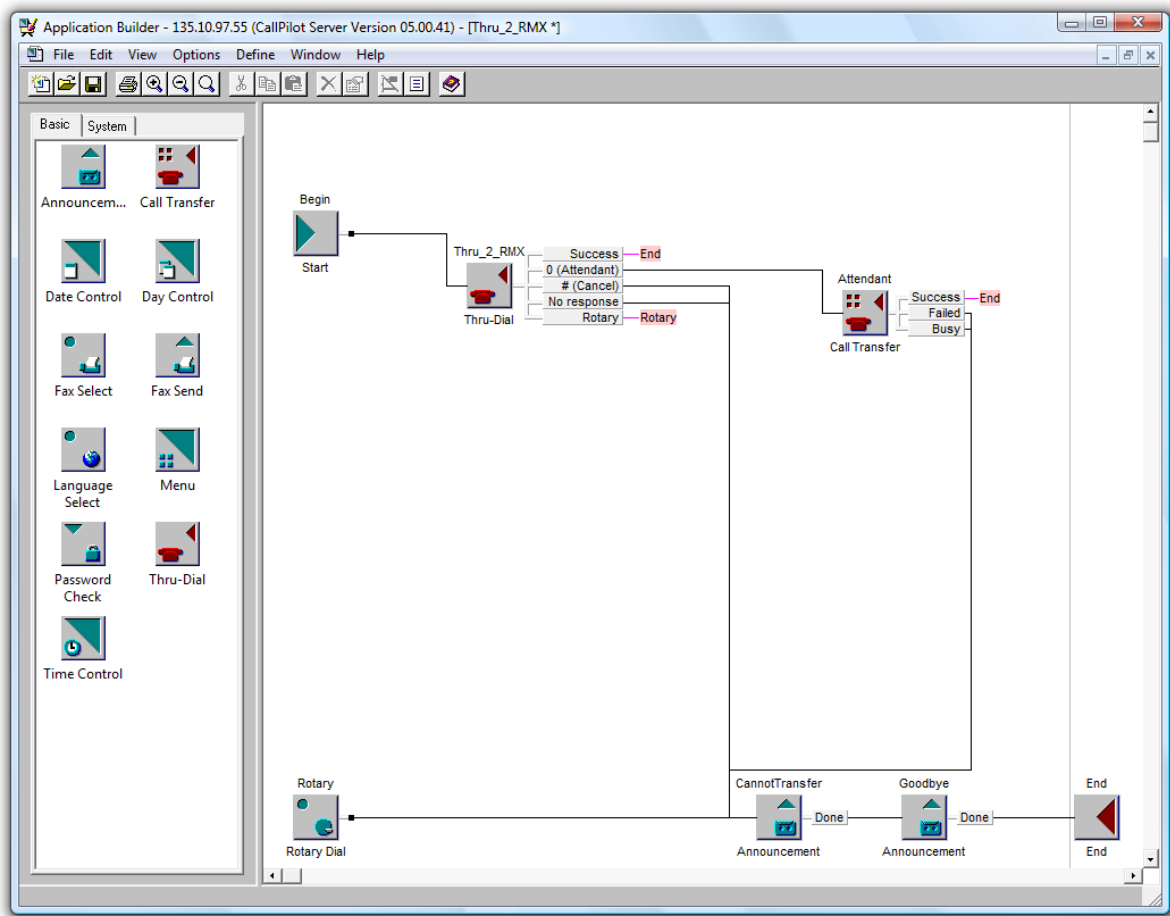


The Thru\_2\_RMX application window is created and opened as shown in **Figure 55** below.



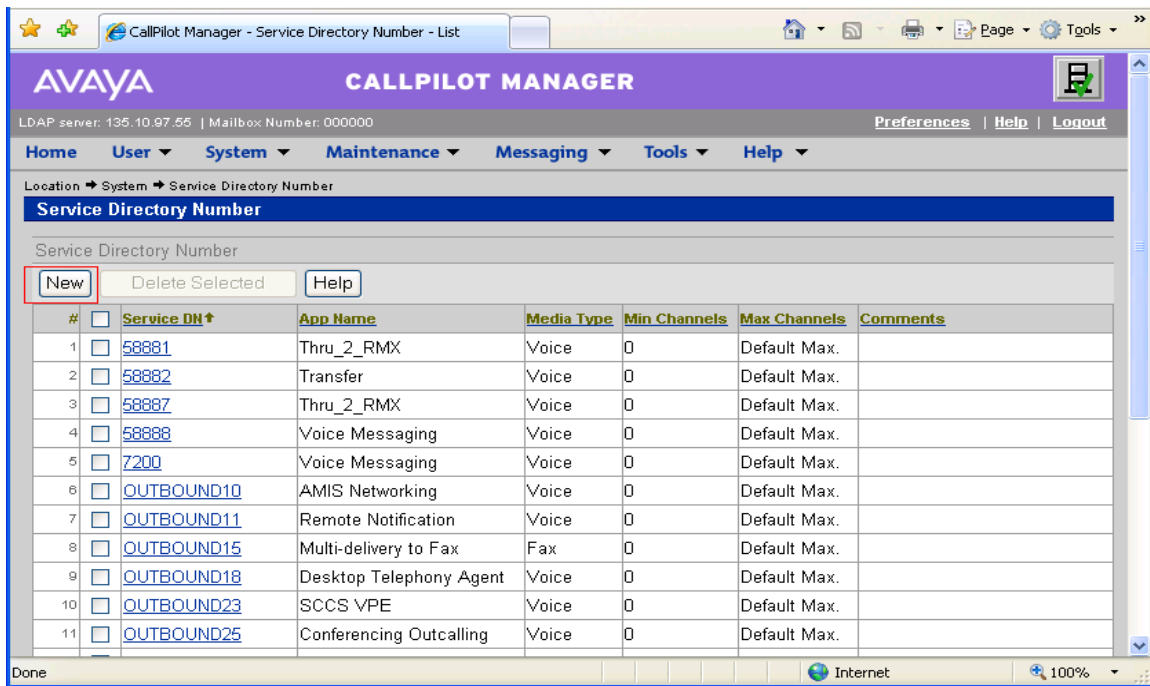
**Figure 55: New application created and ready to build**

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.

The screenshot displays the 'SDN Details' page in the Avaya CallPilot Manager. The browser window title is 'CallPilot Manager - SDN Details'. The page header includes the Avaya logo and 'CALLPILOT MANAGER'. Below the header, there's a navigation bar with links: Home, User, System, Maintenance, Messaging, Tools, and Help. A breadcrumb trail shows 'Location > System > Service Directory Number > SDN Details'. The main content area is titled 'SDN Details:' and contains a 'General' section with the following fields: 'Service DN' (text box with '58881'), 'Application Name' (dropdown menu with 'Thru\_2\_RMX'), 'Media Type' (dropdown menu with 'Voice'), 'Minimum Channels' (text box with '0'), 'Maximum Channels' (checkbox 'Use Default' is checked), 'Remote Activation Password' (text box), 'Password Confirmation' (text box), 'Comments' (text area), and 'Ring-back type' (dropdown menu with 'USA'). At the top of the form, there are buttons for 'Save', 'Cancel', 'Print', and 'Help'. The status bar at the bottom shows 'Done' and 'Internet'.

**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 RMX 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.

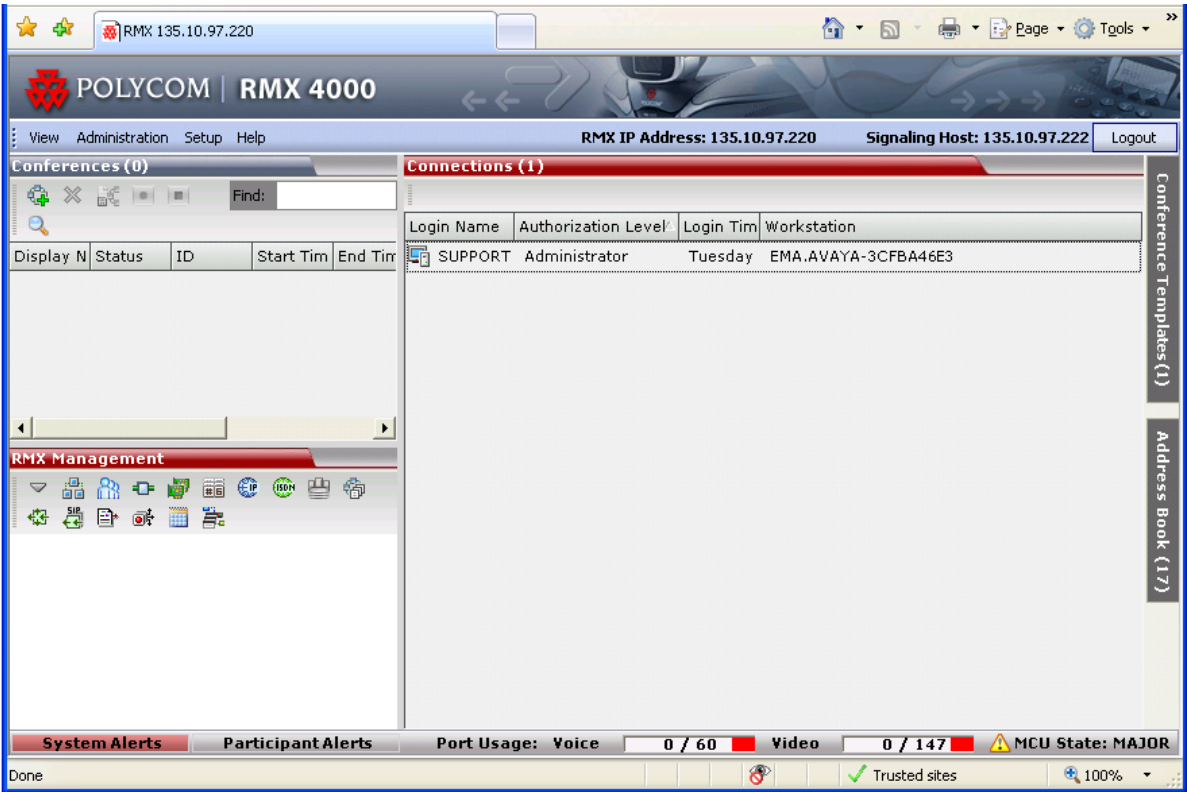
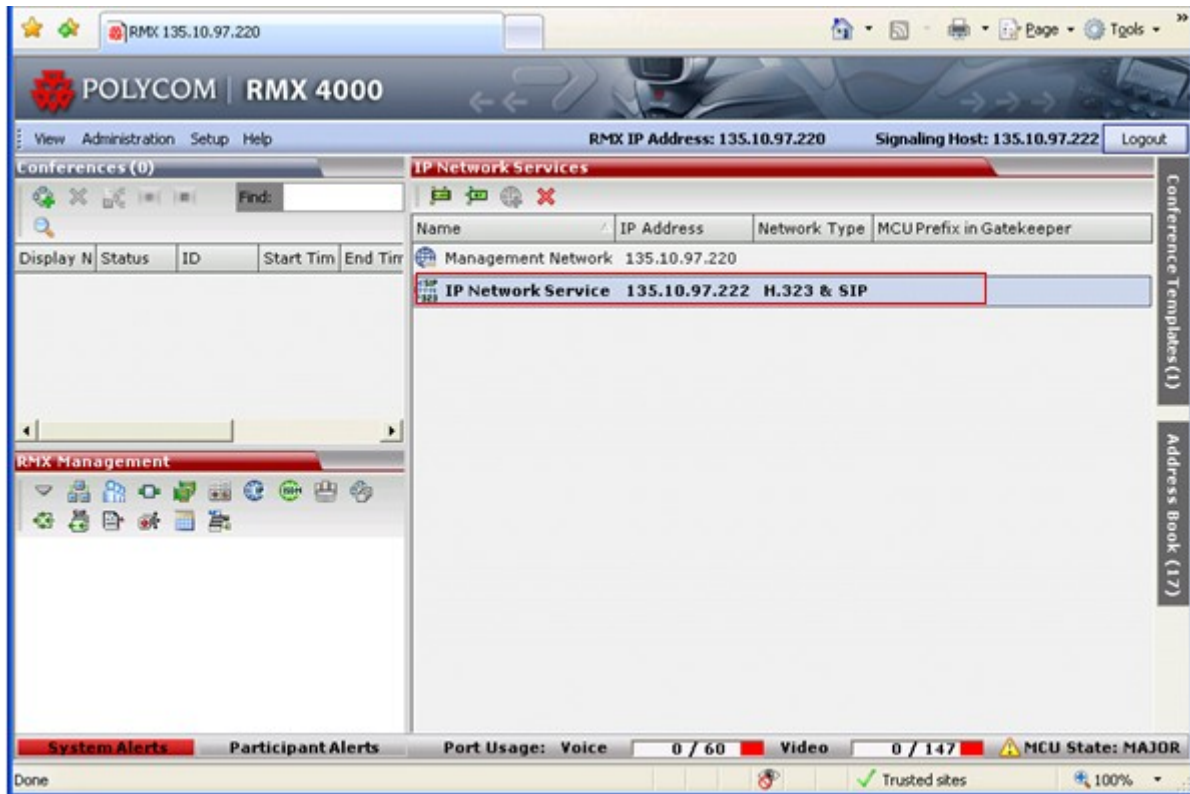


Figure 60: Polycom RMX 4000 Web Client Home Page

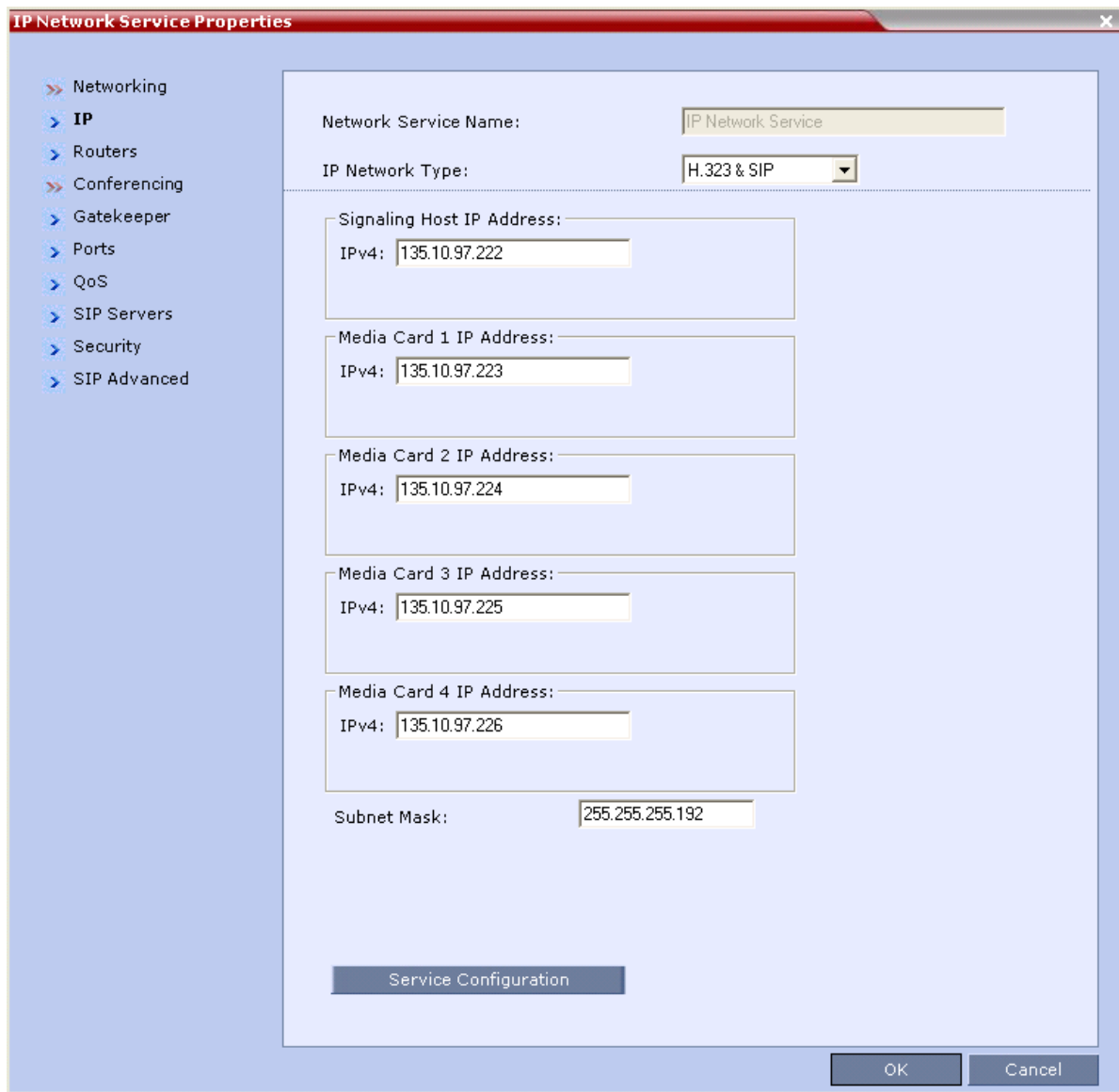
## 8.1. Configure IP Network Service

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**.



The image shows a screenshot of the 'IP Network Service Properties' window. On the left is a tree view with the following items: >> Networking, > IP, > Routers, >> Conferencing, > Gatekeeper, > Ports, > QoS, > SIP Servers, > Security, and > SIP Advanced. The 'IP' item is selected. The main area contains the following fields: 'Network Service Name' with a text box containing 'IP Network Service'; 'IP Network Type' with a dropdown menu showing 'H.323 & SIP'; a section for 'Signaling Host IP Address' with an 'IPv4' field containing '135.10.97.222'; a section for 'Media Card 1 IP Address' with an 'IPv4' field containing '135.10.97.223'; a section for 'Media Card 2 IP Address' with an 'IPv4' field containing '135.10.97.224'; a section for 'Media Card 3 IP Address' with an 'IPv4' field containing '135.10.97.225'; a section for 'Media Card 4 IP Address' with an 'IPv4' field containing '135.10.97.226'; and a 'Subnet Mask' field containing '255.255.255.192'. At the bottom of the main area is a 'Service Configuration' button. At the bottom right of the window are 'OK' and 'Cancel' buttons.

**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 right-hand side as shown in **Figure 63**.

In the **SIP Server** field select as *Specify* and the **Transport 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.

The screenshot shows the 'IP Network Service Properties' window with the 'SIP Servers' tab selected in the left-hand navigation pane. The main configuration area contains the following settings:

- Network Service Name:** IP Network Service
- IP Network Type:** H.323 & SIP
- SIP Server:** Specify
- SIP Server Type:** Generic
- Refresh Registration every:** 3600 seconds
- Transport Type:** UDP
- Certificate Method:** CSR
- Create Certificate** and **Send Certificate** buttons are present.

Below these settings are two tables:

**SIP Servers:**

Parameter	Primary Server	Alternate S
Server IP Address or Name	135.10.97.198	
Server Domain Name	bvwdev.com	
Port	5060	

**Outbound Proxy Servers:**

Parameter	Primary Server
Server IP Address or Name	135.10.97.198
Port	5060

At the bottom right of the window are **OK** and **Cancel** buttons.

**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.

Firstly, select *Specify* in the **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.

The screenshot shows the 'IP Network Service Properties' window with the 'Gatekeeper' tab selected. The configuration is as follows:

- Network Service Name: IP Network Service
- IP Network Type: H.323 & SIP
- Gatekeeper: Specify
- Primary Gatekeeper
  - IP Address or Name: 135.10.97.198
- Backup Gatekeeper
  - IP Address or Name: (empty)
- MCU Prefix in Gatekeeper: 73100
- ☒ Register as Gateway
- Service Mode: board\_hunting
- Refresh Registration every: 120 seconds
- Aliases table:

Alias	Type
	None
	None
	None
	None
	None

**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.

The screenshot shows the 'IP Network Service Properties' dialog box with the 'Gatekeeper' section selected in the left-hand tree. The 'Gatekeeper' dropdown is set to 'Off'. The 'MCU Prefix in Gatekeeper' field contains '73100'. The 'Register as Gateway' checkbox is checked. The 'Service Mode' is 'board\_hunting' and 'Refresh Registration every' is '120 seconds'. An 'Aliases' table is shown at the bottom with five rows, all with 'None' in the 'Type' column.

Alias	Type
	None
	None
	None
	None
	None

**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**.

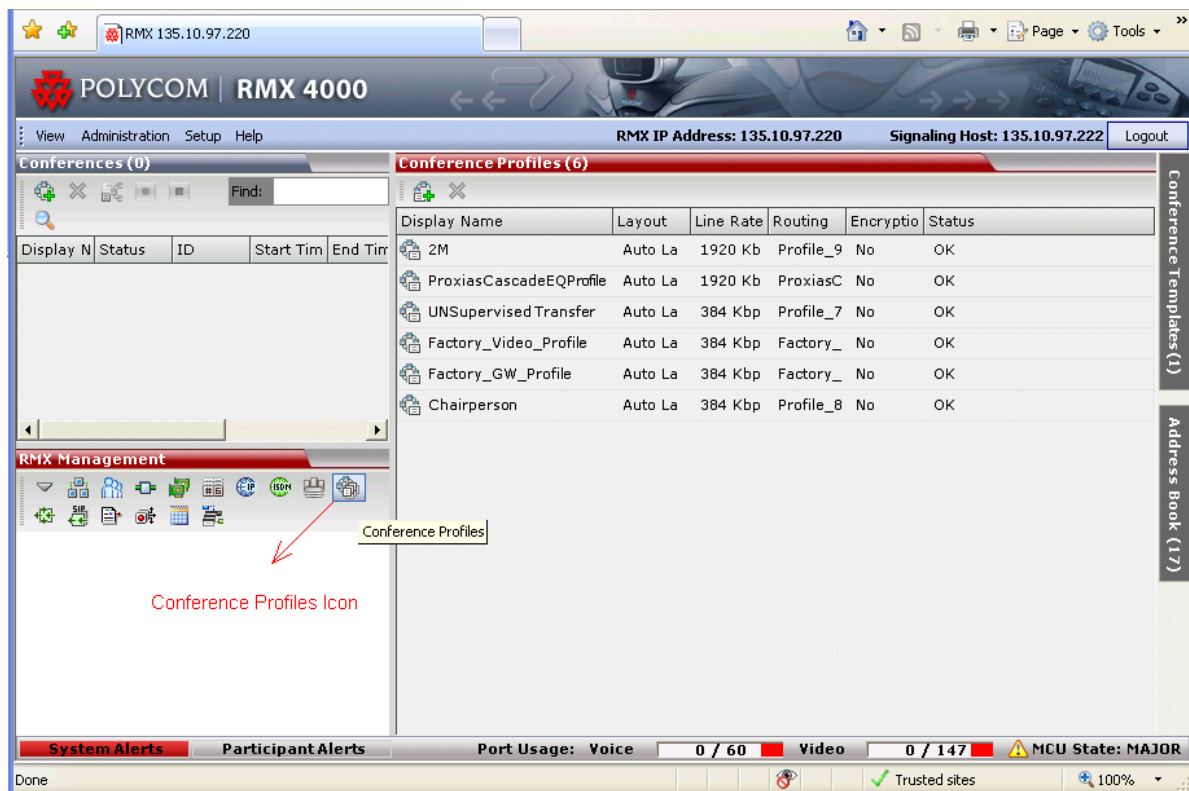
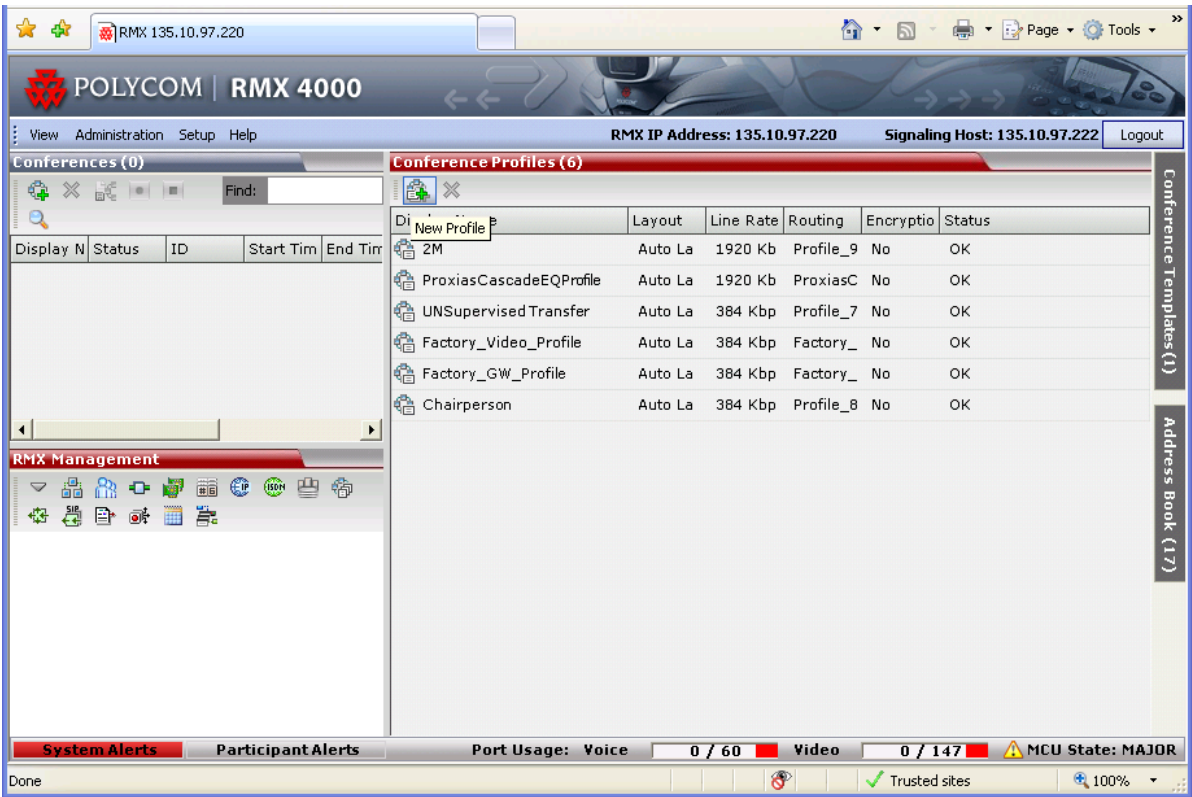


Figure 66: Conference Profiles Window

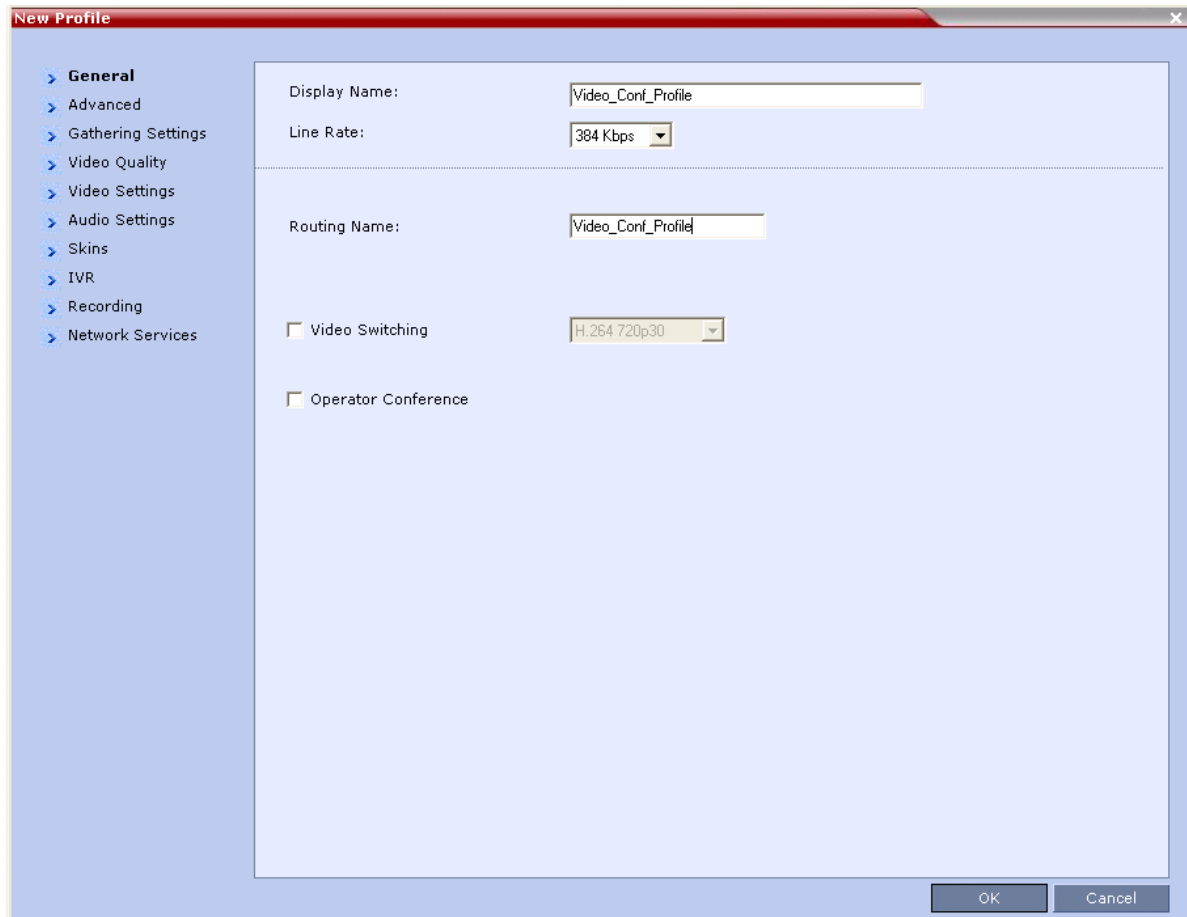
From the right-hand side of the **Conference Profiles** window, click on the **New Profile** icon to create a new conference profile as shown in **Figure 67** below.



**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.



The screenshot shows a window titled "New Profile" with a sidebar on the left containing a tree view of settings categories: General, Advanced, Gathering Settings, Video Quality, Video Settings, Audio Settings, Skins, IVR, Recording, and Network Services. The "General" category is selected. The main area contains the following fields:

- Display Name:** A text box containing "Video\_Conf\_Profile".
- Line Rate:** A dropdown menu showing "384 Kbps".
- Routing Name:** A text box containing "Video\_Conf\_Profile".
- Video Switching:** A checkbox that is unchecked, followed by a dropdown menu showing "H.264 720p30".
- Operator Conference:** A checkbox that is unchecked.

At the bottom right of the window are two buttons: "OK" and "Cancel".

**Figure 68: New Conference Profile window**

Figure 69 below show the new conference profile **Video\_Conf\_Profile** has been created.

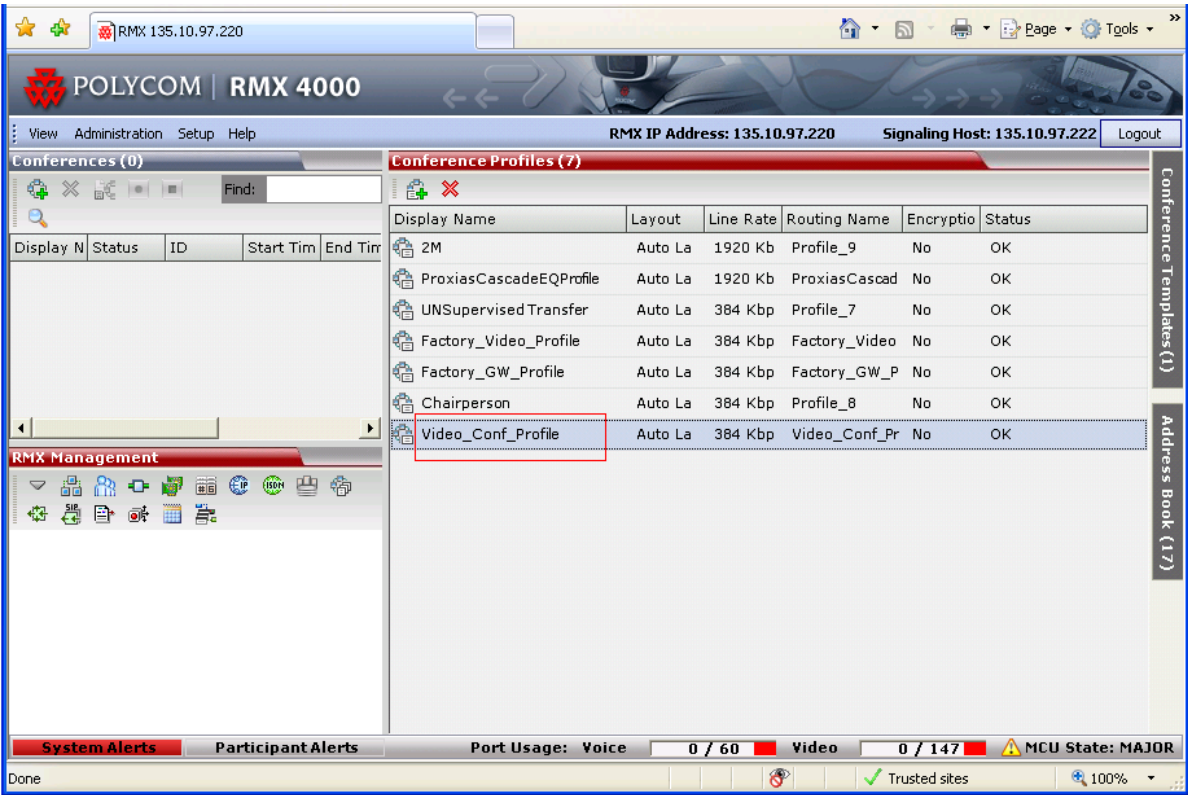
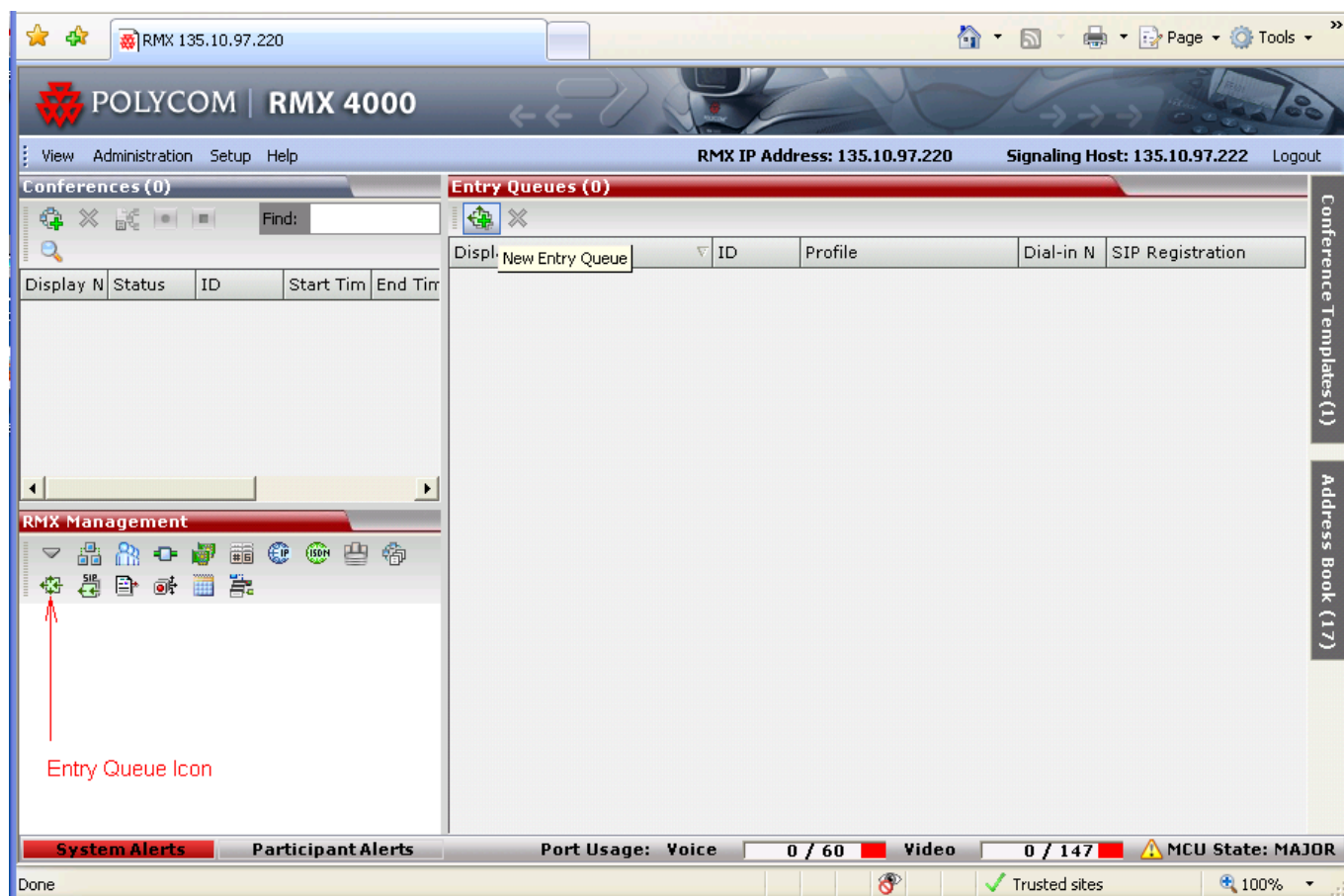


Figure 69: The **Video\_Conf\_Profile** new conference profile created

### 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.

Note: 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.

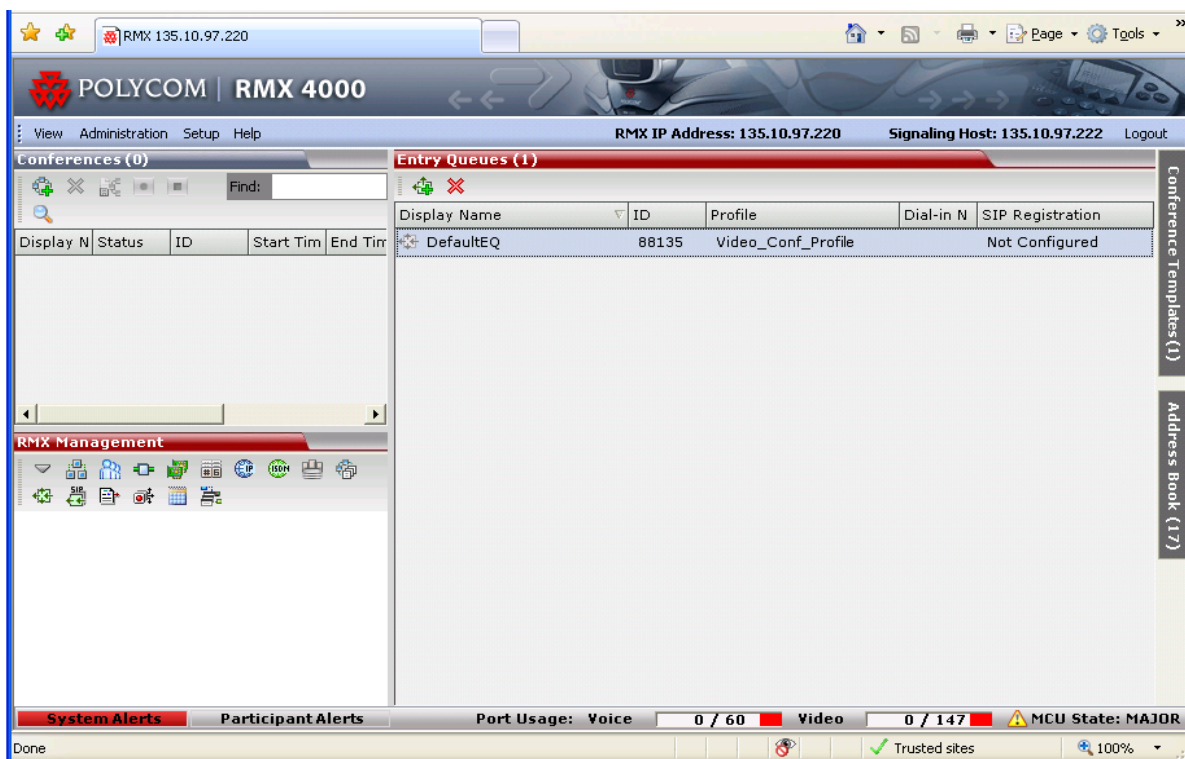
The screenshot shows the 'New Entry Queue' dialog box. The fields are as follows:

- Display Name: DefaultEQ
- Routing Name: DefaultEQ
- Profile: Video\_Conf\_Profile
- ID: (empty)
- Entry Queue IVR Service: Entry Queue IVR Service
- Ad Hoc: ☒
- IVR service provider only: ☐
- Cascade: None
- Enable ISDN/PSTN Dial-in: ☐
- ISDN/PSTN Network Service: [Default Service]
- Dial-in Number (1): (empty)
- Dial-in Number (2): (empty)

Buttons: OK, Cancel

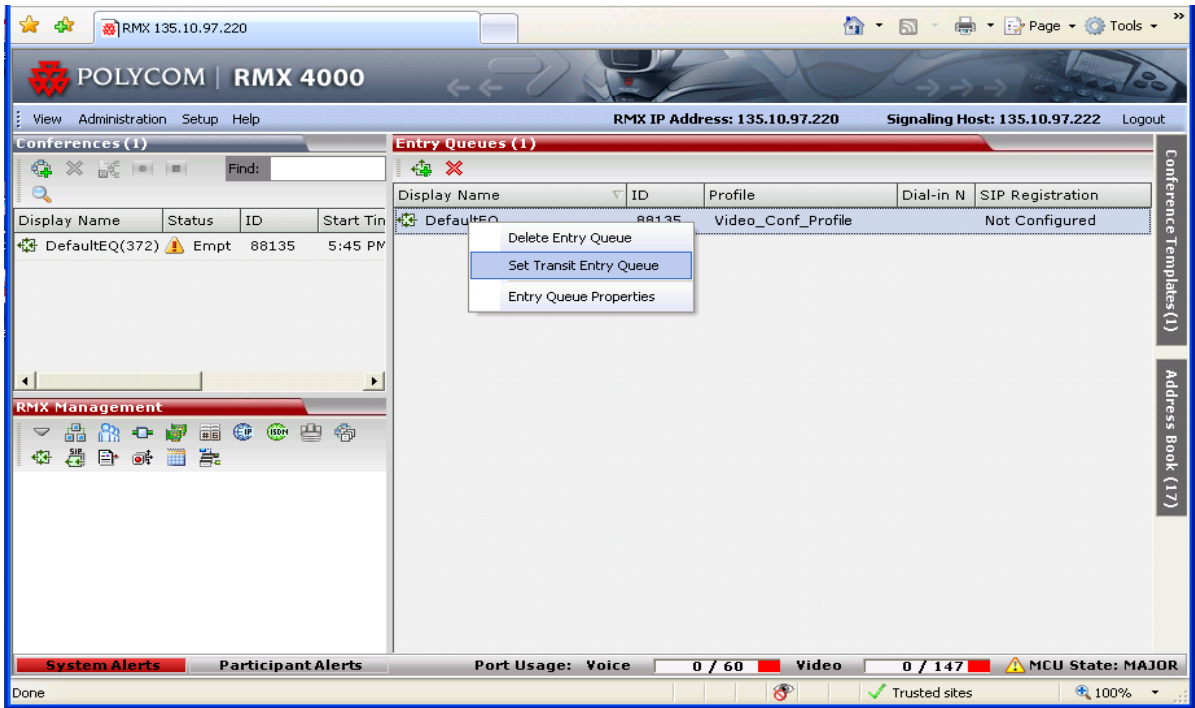
**Figure 70: New Entry Queue Window**

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**.

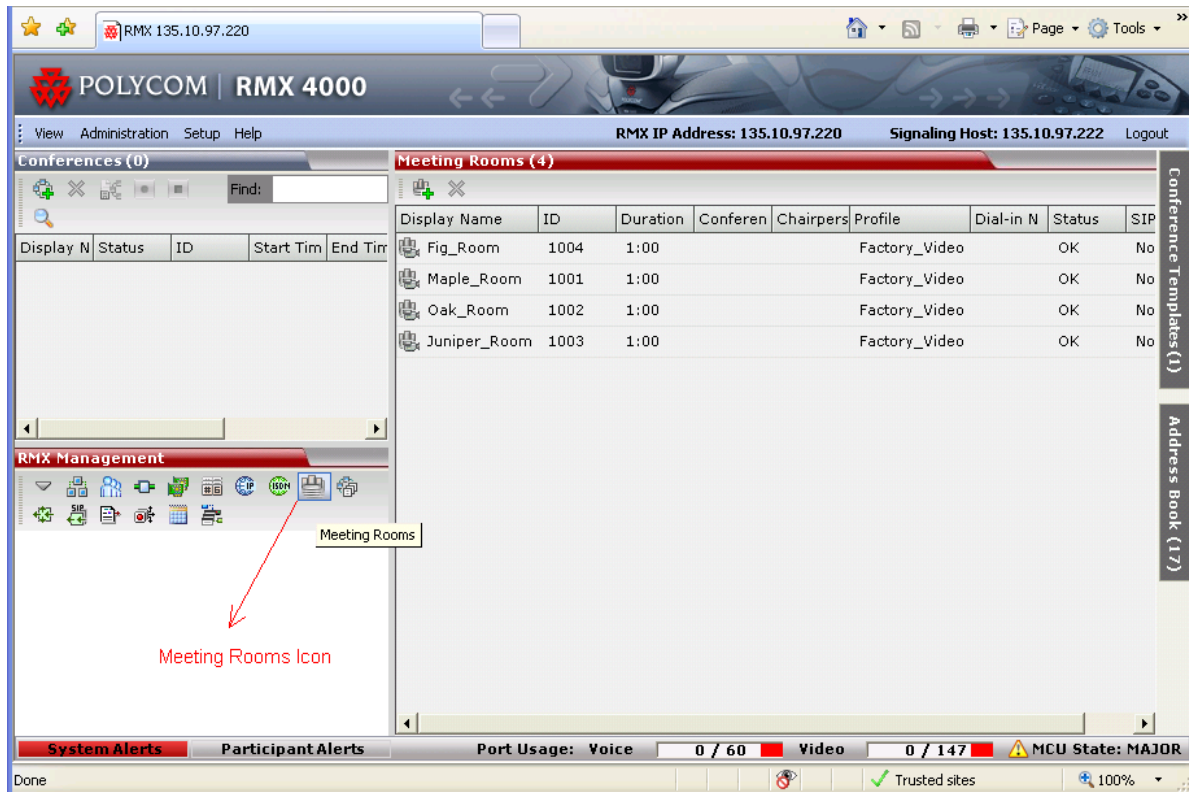


Figure 73: The Meeting Rooms window

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 *Belleville\_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.

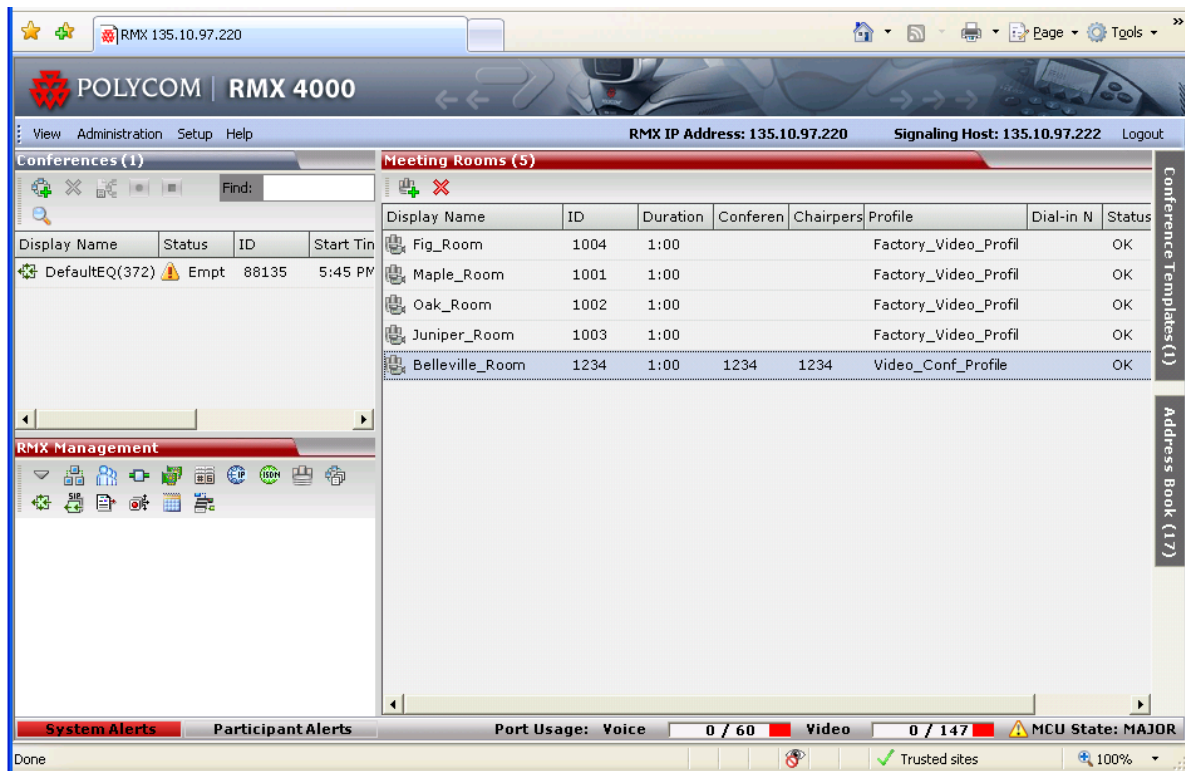
The screenshot shows a window titled "New Meeting Room" with a sidebar on the left containing three expandable sections: "General", "Participants", and "Information". The "General" section is currently expanded. The main area contains the following fields and controls:

- Display Name:** Text box containing "Belleville\_Room".
- Duration:** Spinners for hours (set to 1) and minutes (set to 00), followed by a checkbox labeled "Permanent Conference".
- Routing Name:** Text box containing "Belleville\_Room".
- Profile:** Dropdown menu showing "Video\_Conf\_Profile".
- ID:** Text box containing "1234".
- Conference Password:** Text box containing "1234".
- Chairperson Password:** Text box containing "1234".
- Reserve Resources for Video Participants:** Spinner set to "0".
- Reserve Resources for Voice Participants:** Spinner set to "0".
- Maximum Number of Participants:** Dropdown menu set to "Automatic".
- Enable ISDN/PSTN Dial-in:** A checkbox that is currently unchecked.
- ISDN/PSTN Network Service:** Dropdown menu showing "[Default Service]".
- Dial-in Number (1):** Empty text box.
- Dial-in Number (2):** Empty text box.

At the bottom right of the window are "OK" and "Cancel" buttons.

**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 DN.
- 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

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