



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

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# **Application Notes for Movitas Hosted Solution over SIP Trunk between Movitas MvPBX System and Avaya Communication Server 1000 Release 7.5 – Issue 1.0**

### **Abstract**

These Application Notes describe a solution comprised of SIP Trunk interoperating between Movitas MvPBX System and Avaya Communication Server 1000 Release 7.5.

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 the interoperability of Avaya's CS1000 Enterprise PBX and Movitas's MvPBX system and its respective iPhone, Android and Blackberry applications. The testing will include multiple call scenarios including calls between desk phones and Movitas application users and calls to outside PSTN lines through the CS1000 system.

## 2. General Test Approach and Test Results

The General test approach was to verify the SIP interoperability between both Avaya CS1000 Enterprise PBX and Movitas MvPBX systems as outlined in Section 2.1.

During the compliance test there were 3 extensions of CS1000 named as Front Desk, Room 1 and Room 2; these extensions were associated with 3 respective SIP Users in the Movitas system. The Movitas SIP User is a part of Movitas Dreams Digital Cancun application installed in smart mobile devices including iPhone, Android and Blackberry. These SIP Users are registered to the Movitas MvPBX system. Any call that comes in to one of the three CS1000 extensions will also ring their associated SIP User in the smart phone. The call can be accepted either on the extension or the SIP User. Whichever phone answers the call, the other stops ringing and becomes idle.

### 2.1. Interoperability Compliance Testing

The focus of this testing was to verify the SIP Trunk interoperability in between Communication Server 1000 and Movitas MvPBX in placing multiple calls between two systems. The following test areas were practiced in the compliance testing:

- Verification of SIP Trunk registration of CS1000 SIP Signaling gateway to Movitas SIP MvPBX
- Verification of call from Movitas SIP User to a CS1000 extension.
- Verification of call from CS1000 extension to a Movitas SIP User.
- Verification of call from Movitas SIP User to PSTN over SIP Trunk.
- Verification of SIP settings compatibility for codec support and packet size negotiation.

### 2.2. Test Results

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

There is a pending issue of SIP Trunk registration between CS1000 SIP Gateway and Movitas SIP MvPBX as the SIP Trunk registration keeps disconnecting after period of time. This issue is being investigated from both Avaya and Movitas; wi00940727 has been filed to track the issue.

### 2.3. Support

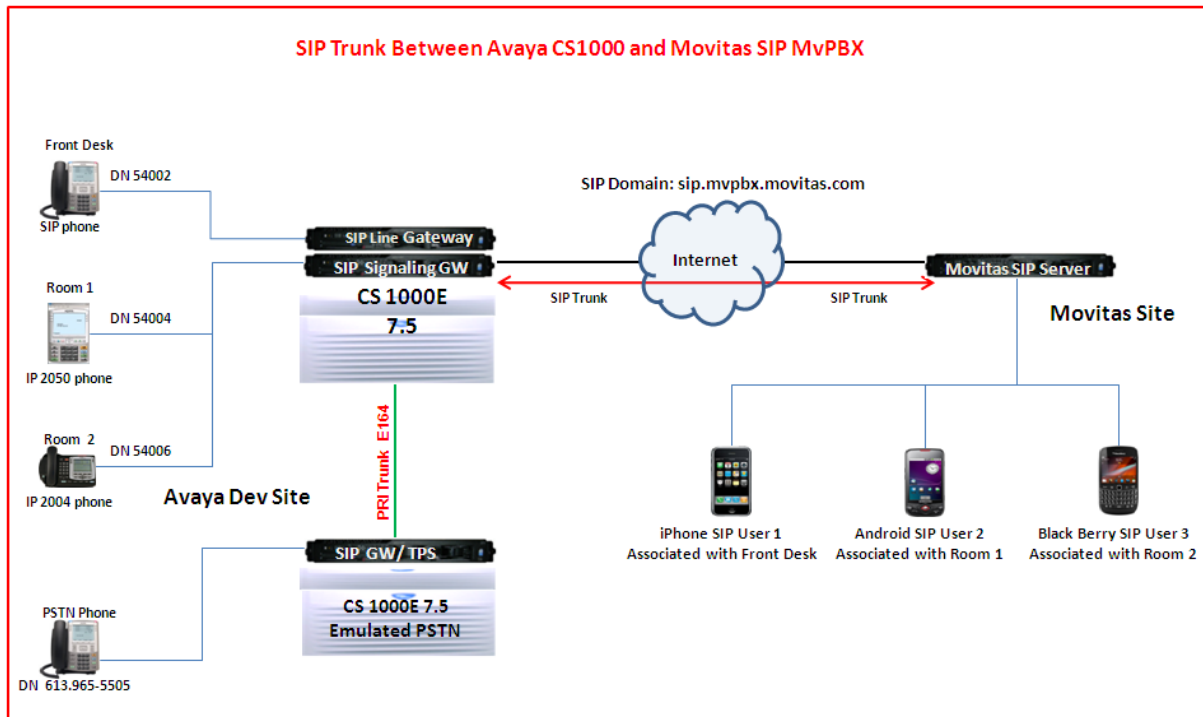
For technical support for the Movitas Hosted Solution, and Movitas products in general, please refer to [www.movitas.com](http://www.movitas.com).

Telephone: 888-343-3721 ext. 1

Email: [support@movitas.com](mailto:support@movitas.com)

### 3. Reference Configuration

**Figure 1** illustrates the network diagram configuration used during the compliance testing between the Avaya Communication Server 1000 and Movitas SIP MvPBX system, the second CS 1000 system is used to simulate PSTN calls.



**Figure 1: Network Diagram 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 DepList 1: core Issue: 01
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
iPhone Smart Phone OS	iPhone OS version 4.3.5
Android Smart Phone OS	DroidX OS version 2.3.3
Blackberry Smart Phone OS	Bold 9700 OS version 6.0.0.448
Movitas MvPBX Build	i-481ec228
iOS Movitas Phone app using Test Dreams as a client	1.2
Android Dreams Application app using Test Dreams as a client	2.0
Blackberry Movitas Phone app using Test Dreams as a client	1.0

Note: The Movitas Phone application with Test Dreams was used in replacement of the planned Dreams Cancun app to prevent downtime with the telephony component of the live Dreams Cancun app.

## 5. Configure Avaya Communication Server 1000

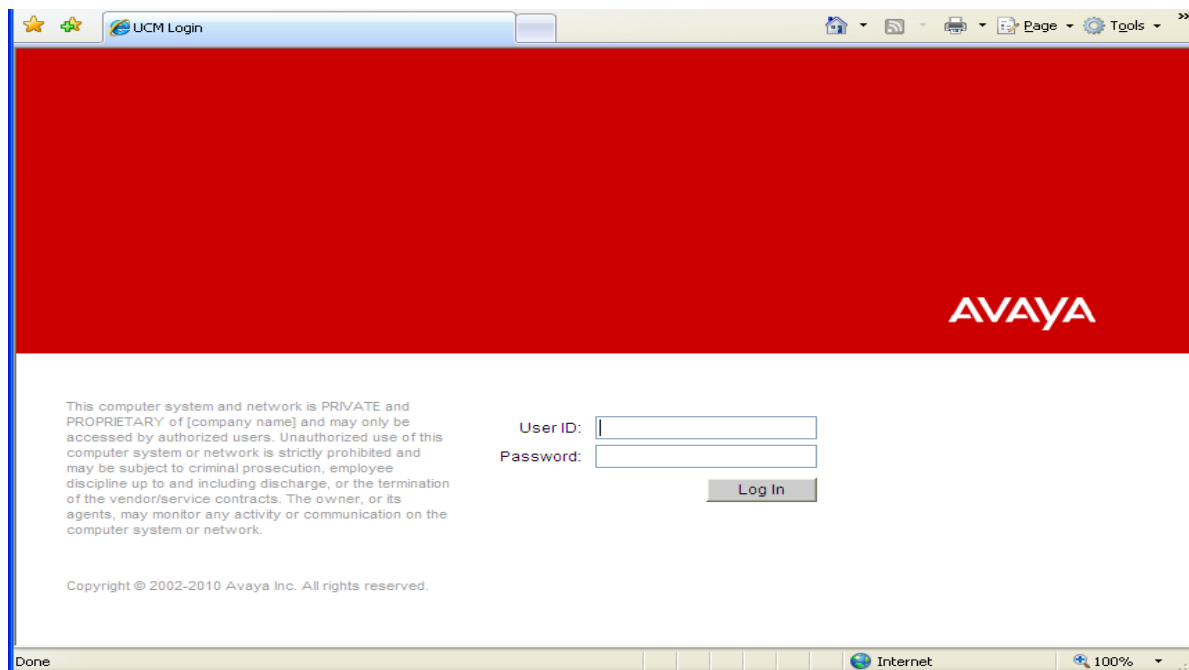
This document assumes that the Avaya Communication Sever 1000 system was properly installed and configured as per the product documents. This section provides the steps on how to provision the CS1000 to work with the Movitas MvPBX system. For more information about how to install and configure Communication Sever 1000, please refer to **Section 9 [1]**.

The following summarizes the tasks which need to be done on the CS1000 System:

- Register the CS1000 SIP Signaling Gateway to Movitas MvPBX.
- Configure D-Channel for SIP Trunk
- Configure Zone for Route and Trunk.
- Configure SIP Route.
- Configure SIP Trunks.
- Configure CDP Dialing plan.
- Configure IP Phone and its associated PCA

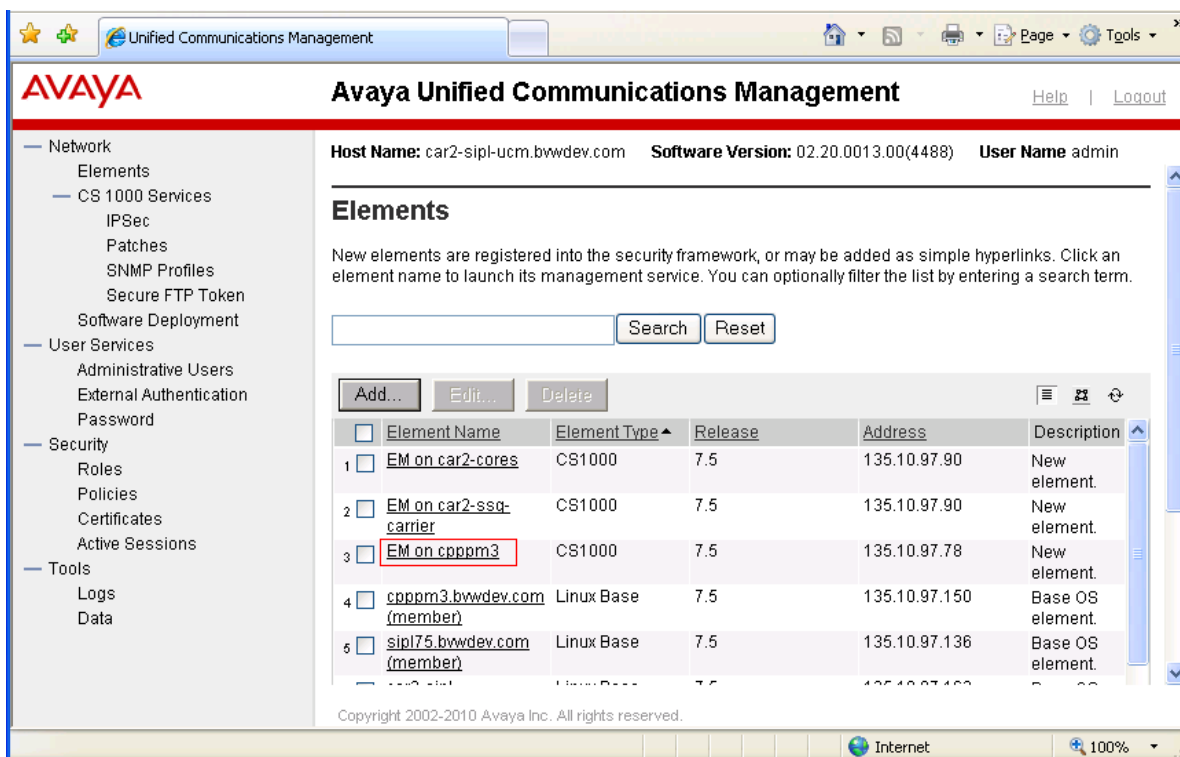
### 5.1. Register the CS1000 SIP Signaling Gateway to Movitas MvPBX

To register the CS1000 SIP Signaling Gateway to the Movitas MvPBX, 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 2.



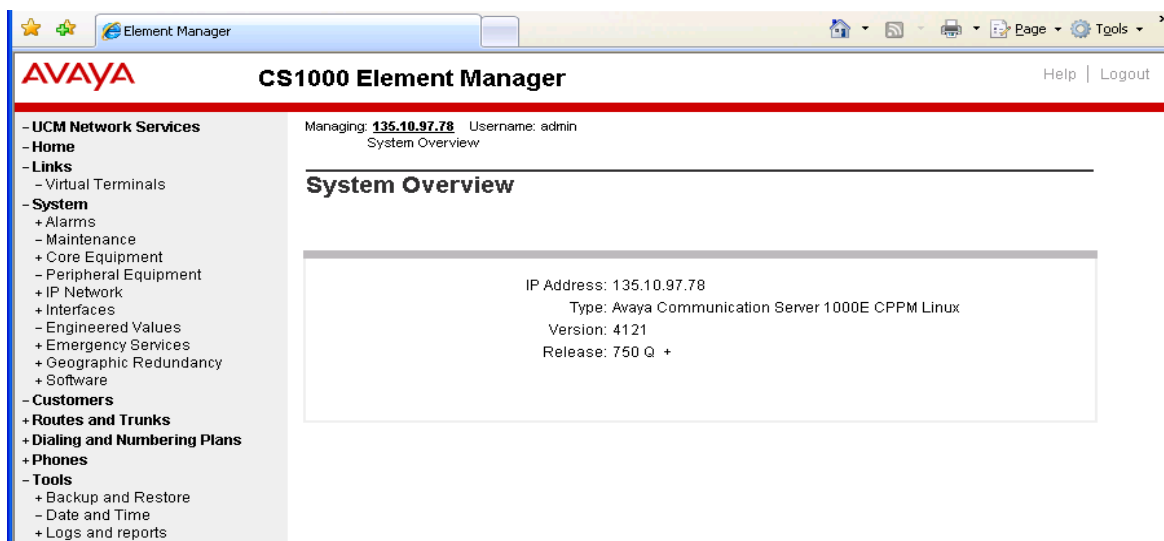
**Figure 2: UCM Login window**

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 3.



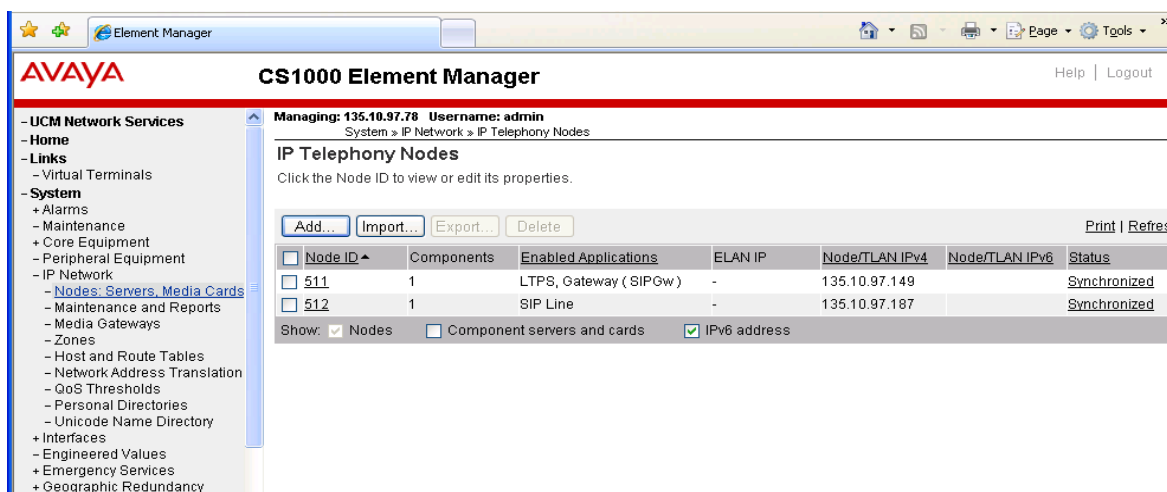
**Figure 3: UCM Home Page**

Click on the **Element Name** link, in this sample is “EM on cpppm3”, that manages the CS1000 system, the Element Manager window appears as shown in Figure 4.



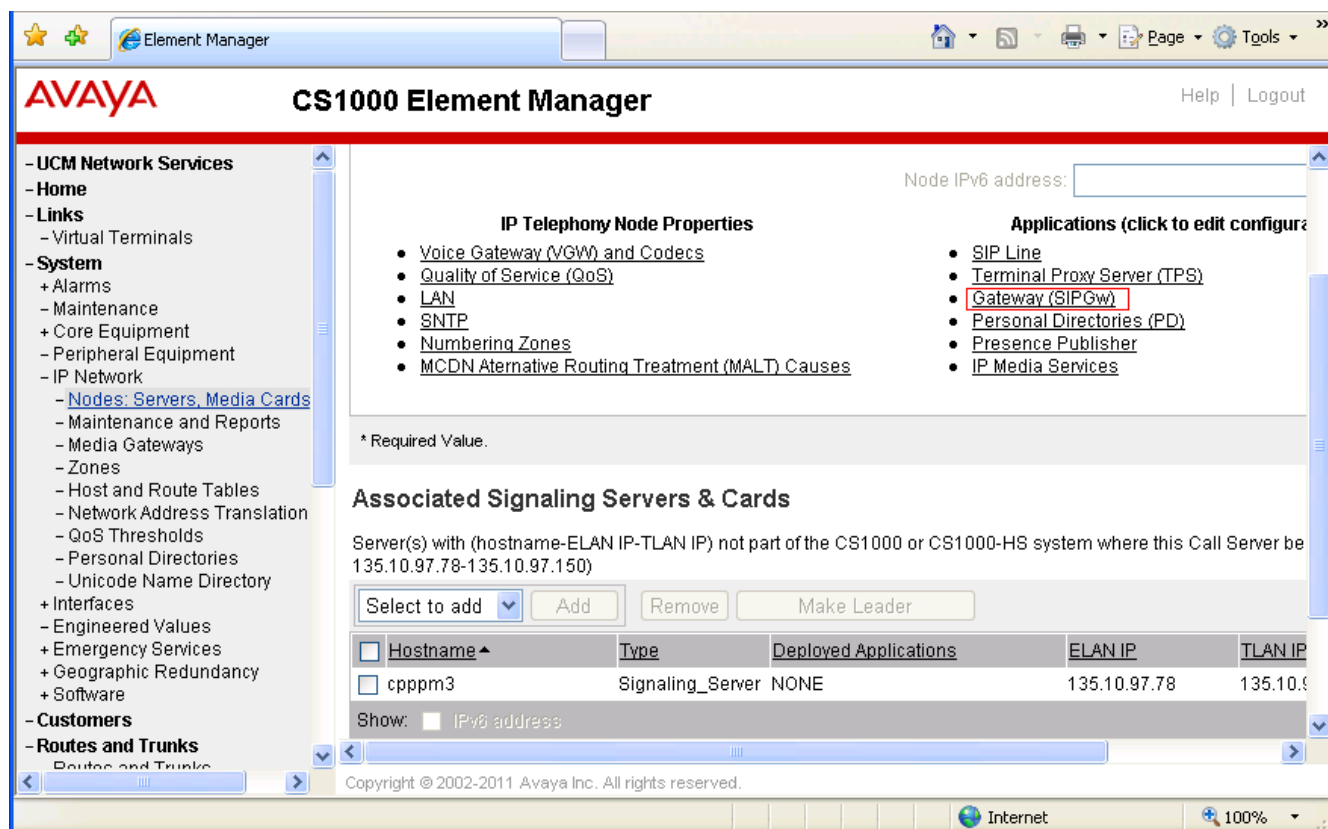
**Figure 4: CS1000 Element Manager Home page**

On left-hand side of the Element Manager window and under the **System** tab, expand the **IP Network > Nodes: Servers and Media Cards**, the **IP Telephony Nodes** is displayed in the right-hand side of the window as shown in Figure 5.



**Figure 5: IP Telephony Nodes Page**

Click on the **Node ID**, in this sample is **511**, which has the **SIPGw** application enabled; the Node **511** detail appears as shown in Figure 6.



**Figure 6: IP Telephony Node Detail Page**

Under the **Applications**, click on the **Gateway (SIPGw)** application link, the **Node ID: 511 – Virtual Trunk Gateway Configuration Details** appears, in the **General** section, enter the domain *sip.mypbx.movitas.com* in the **SIP Domain Name** box and **Local SIP Port 5060**, **Gateway Endpoint Name** *avayatest* & password in the **Gateway password** and **Application Node ID** as **511** as shown in Figure 7.

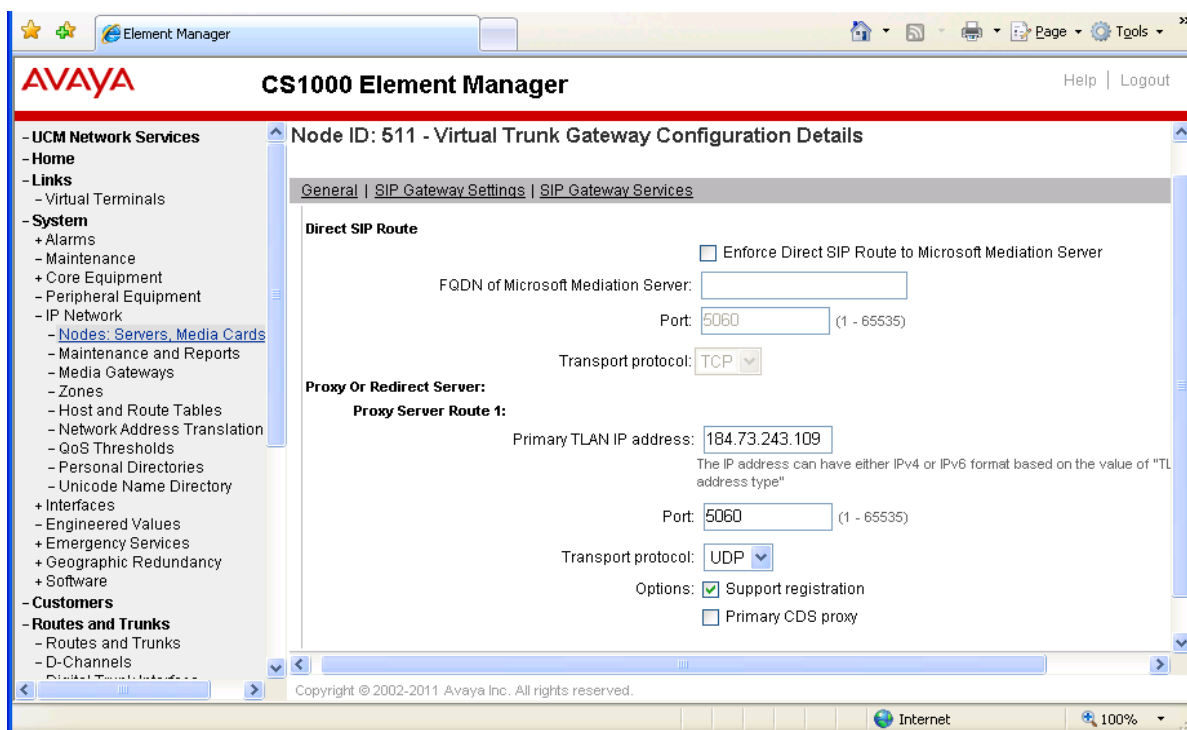
Note: The Gateway Endpoint name and password was given by Movitas and it should be matched with configuration on the Movitas MvPBX system.

The screenshot displays the AVAYA CS1000 Element Manager web interface. The left sidebar shows a navigation tree with categories like UCM Network Services, Home, Links, System, and Customers. The main content area is titled 'General | SIP Gateway Settings | SIP Gateway Services'. It contains several input fields: 'Vtrk gateway application' (SIP Gateway (SIPGw)), 'SIP domain name' (sip.mypbx.movitas.com), 'Local SIP port' (5060), 'Gateway endpoint name' (avayatest), 'Gateway password' (masked with dots), and 'Application node ID' (511). There are also checkboxes for 'Monitor IP addresses' and 'Enable failsafe NRS', and radio buttons for 'SIP ANAT' (IPv4/IPv6). A 'Save' button is at the bottom right.

**Figure 7: Node ID: 511 – Virtual Trunk Gateway Configuration Details**

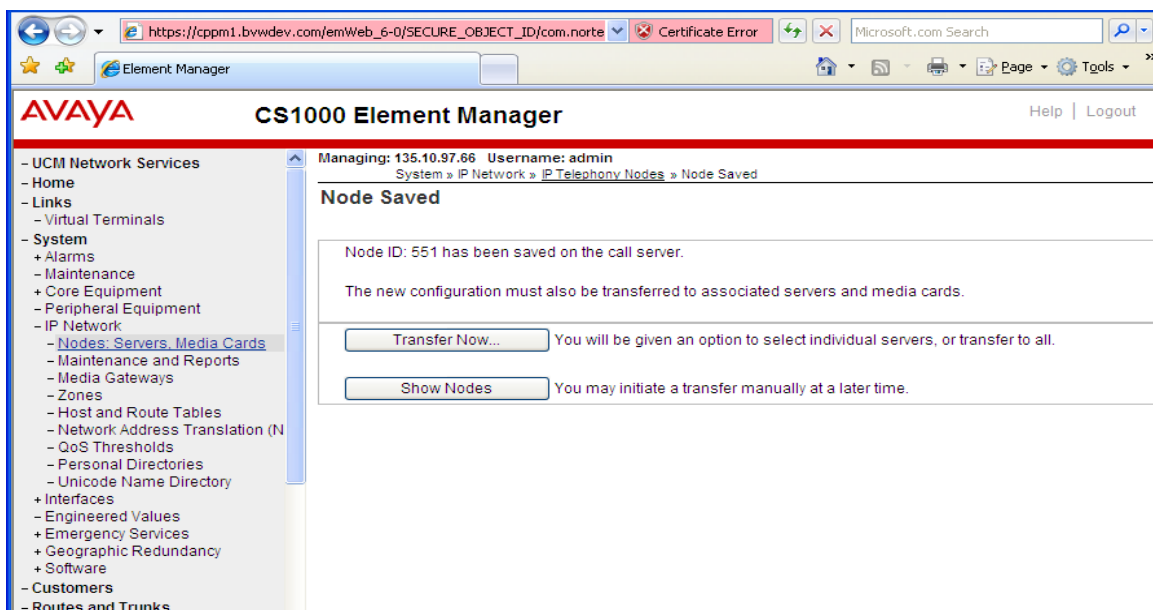


Continue scrolling down to the section **SIP Gateway Settings**, in the **Proxy Or Redirect Server** of this section, enter the IP address of Movitas MvPBX in the field **Primary TLAN IP address**, **Port 5060**, **Transport UDP** and check in the option **Support registration** as shown in Figure 8.



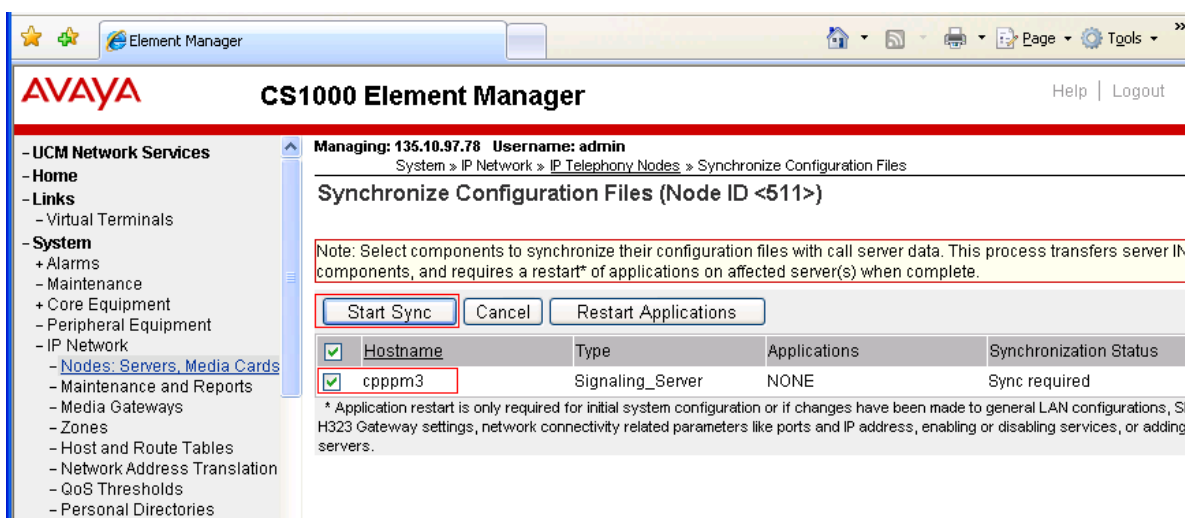
**Figure 8: The SIP Gateway Setting of Telephony Node**

Click on the **Save** button at the bottom of this page (not shown) to save the changes in the Node ID 551, the **Node ID 511 - Virtual Trunk Gateway Configuration Detail** window will be closed and return back to the **Node Details (ID: 511 - LTPS, Gateway (SIPGw))** page. Click on the **Save** button in this page and the **Node Saved** window appears as shown in Figure 9.



**Figure 9: Node Saved Page**

Click on the **Transfer Now...** button in Figure 9 and the **Synchronize Configuration Files (Node ID)** page appears as shown in Figure 10. Click on the associated server, in this sample is **cpppm3**, and click on the **Start Sync** button to start transferring the changes to this server.



**Figure 10: Synchronize Configuration Files (Node ID <551>) page**

## 5.2. Configure D-Channel for SIP Trunk

To configure a D-Channel for SIP 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 11.

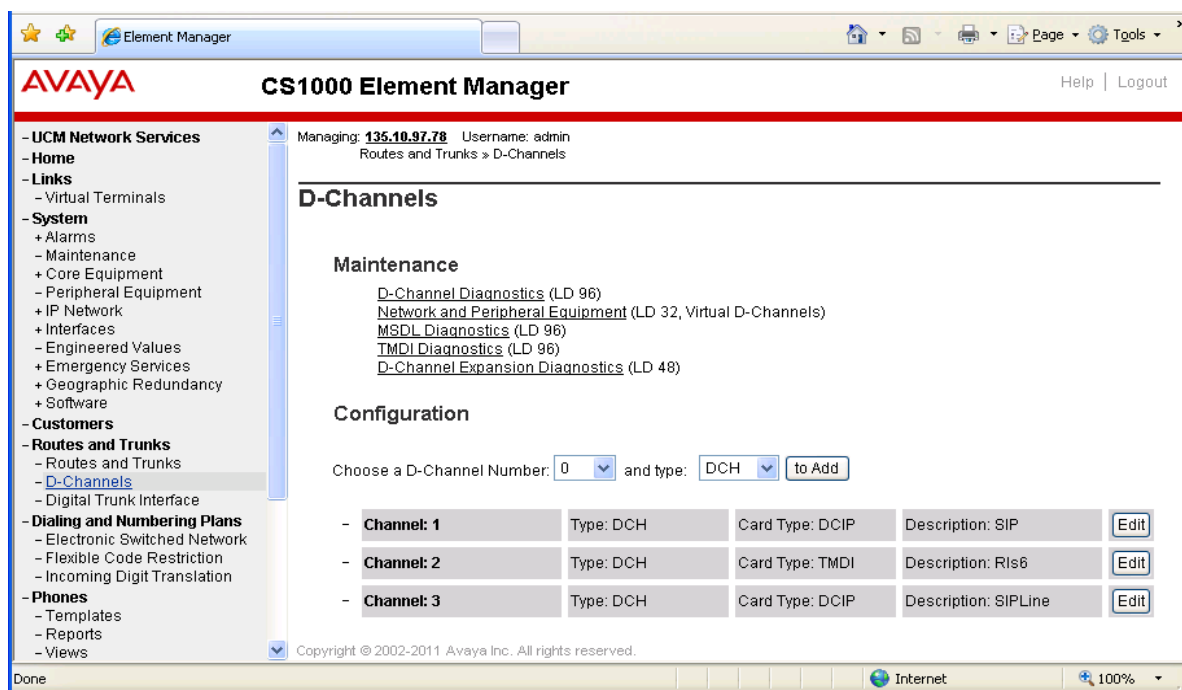


Figure 11: D-Channels page

In the **Configuration** section of this page, select an available D-Channel in the **Choose a D-Channels Number** dropdown list, select the type of D-Channel as **DCH** and click on the **Add** button.

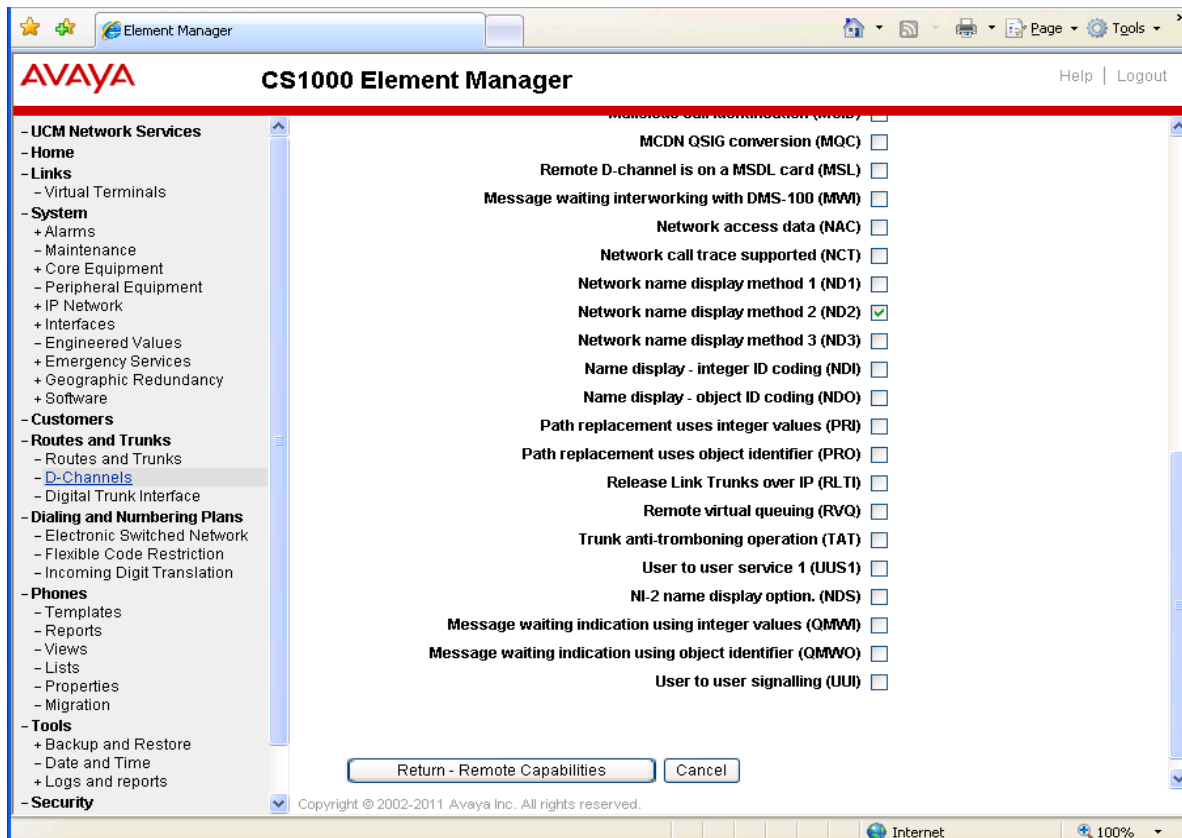
The **Basic Configuration** of new D-Channel appears as shown in Figure 12. Select **D-Channel is over IP (DCIP)** in the **D-Channel Card Type**, enter a description in the **Designator** box and keep all other values at their defaults.

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 12: 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 and the **Remote Capabilities Configuration** page appears as shown in Figure 13. Check on the checkbox of **Network name displayed method 2 (ND2)** and click on **Return – Remotes Capabilities** button to go back to the **Basic options** section. Keep all other values of this section at their defaults.



**Figure 13: Remote Capability page**

Continue expanding the **Advanced options** subsection of the **Basic Configuration** section. The **Advanced options** section appears as shown in Figure 14 below. Keep all remaining values at their defaults.

**Figure 14: Advanced options (ADVOPT)**

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

### 5.3. Configure Zone Bandwidth

To configure a Zone, 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 15.

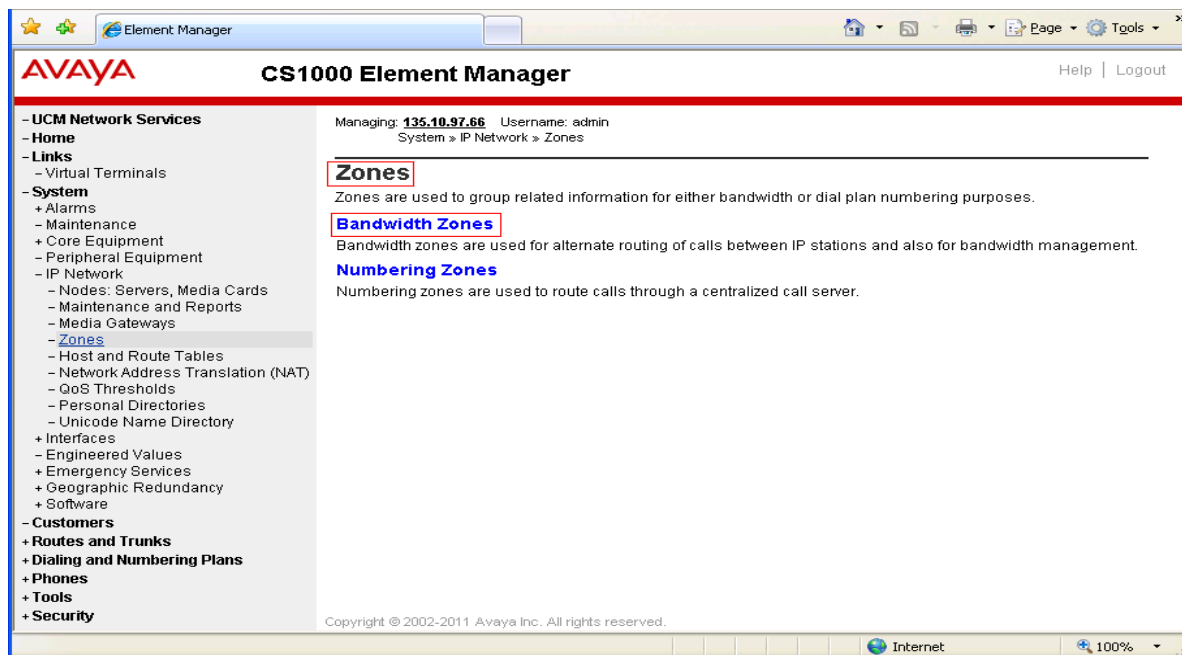


Figure 15: Zones Page

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. The **Zone Basic Property and Bandwidth Management** page appears as shown in Figure 16. Enter **4** in the **Zone Number**, select **Zone Intent (ZBRN)** as **VTRK** (because this zone is used for virtual trunks) and keep other fields at their defaults. Click on **Save** button to save changes and complete adding the new zone.

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 - 100000000 )
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ) ▼
Interzone Bandwidth (INTER_BW):	1000000 ( 0 - 100000000 )
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 16: Zone Basic Property and Bandwidth Management



## 5.4. Configure SIP Route

To configure a SIP Route from the homepage of Element Manager, navigate to **Routes and Trunks > Routes and Trunks**. The **Routes and Trunks** page appears in the right-hand side as shown in Figure 17.

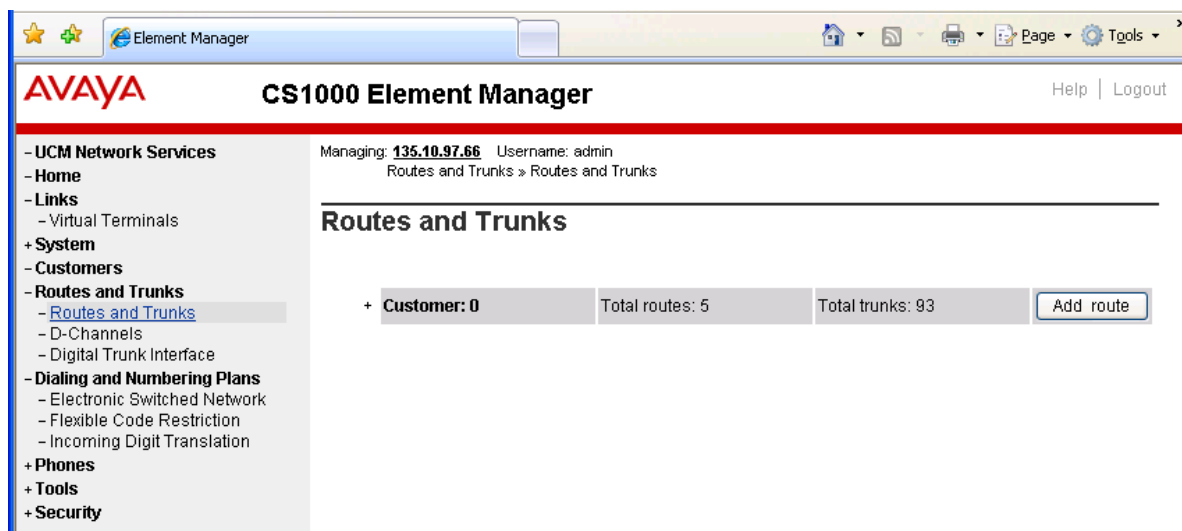


Figure 17: Routes and Trunks page

Identify the customer to which the new route is going to be added (in this sample there is just one, Customer 0) and then click on the **Add route** button. The **New Route Configuration** page appears as shown in Figure 18 and consists of 5 sections: **Basic Configuration**, **Basic Route Options**, **Network Options**, **General Options**, and **Advanced Configurations**.

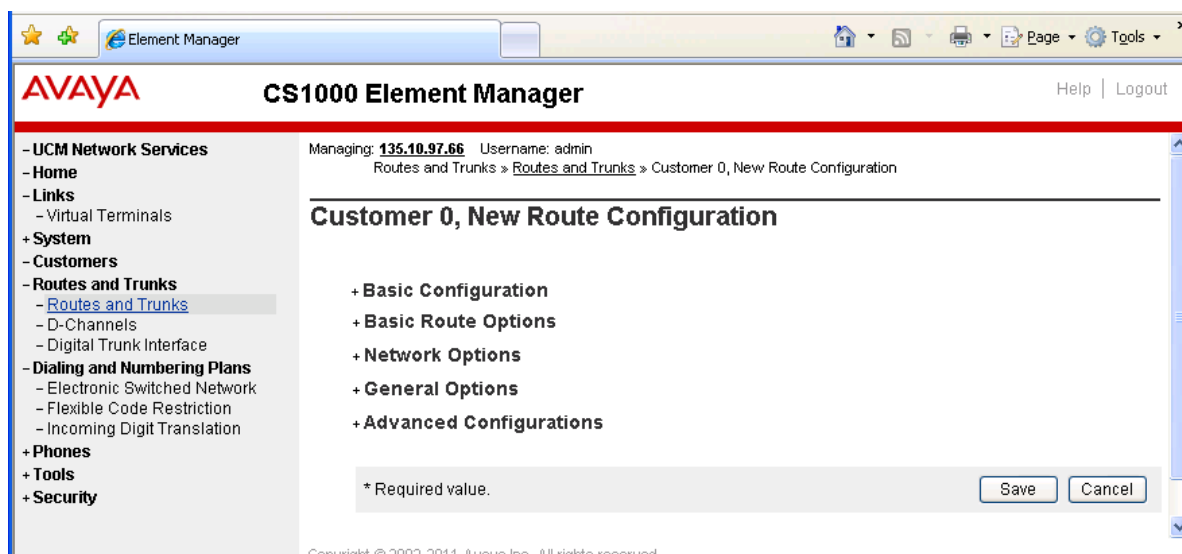


Figure 18: New Route Configuration page

Figure 19 below shows the **Basic Configuration** section with values entered as typical for 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 ID for codec selection and bandwidth management (ZONE):** 4 -> as defined in the Section 5.3
- **Node ID of signaling server of this route (NODE):** 551 -> This Node is used to register to the Session Manager in the Section 5.1
- **Calling number dialing plan (CPND):** 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 area is titled 'Basic Configuration' and contains the following fields:

- 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 M811P (M811P): ☐
- 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 main configuration fields are sections for '+ Basic Route Options', '+ Network Options', '+ General Options', and '+ Advanced Configurations'. The footer of the window shows 'Copyright © 2002-2011 Avaya Inc. All rights reserved.' and a status bar with 'Internet' and '100%' zoom.

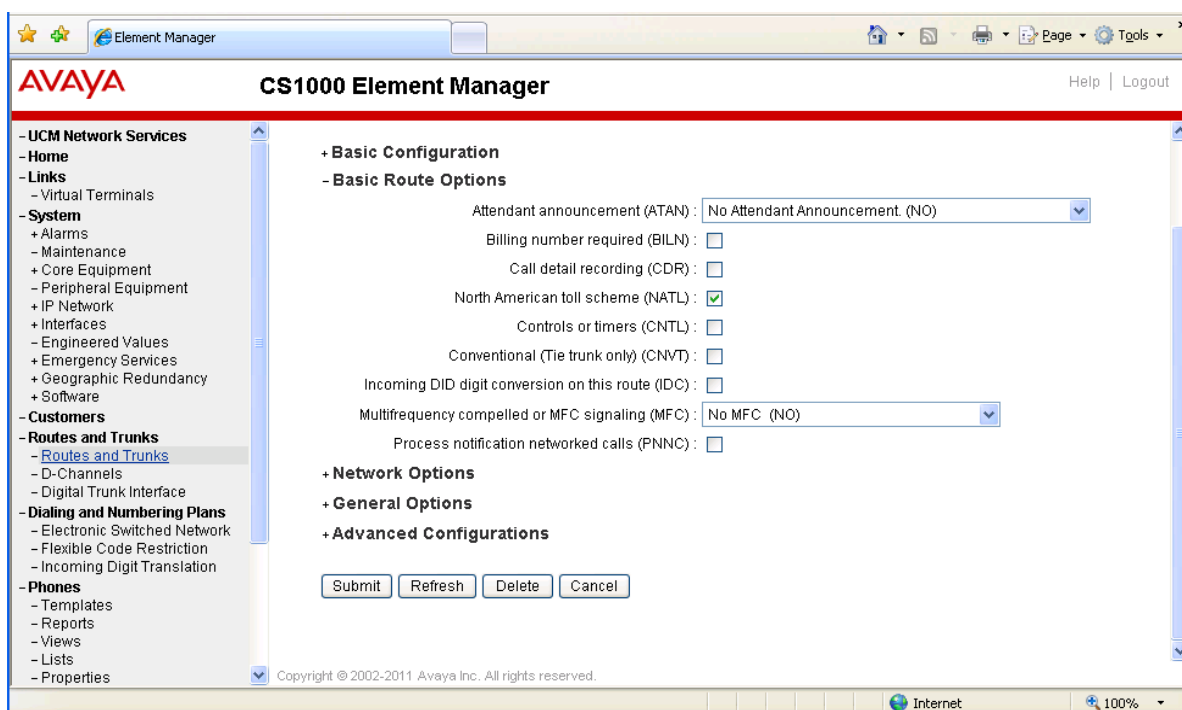
**Figure 19: Basic Configuration section of new route**

Checked on the **Integrated services digital network option (ISDN)** in the **Basic Configuration** section. Figure 20 below shows the sub-options for this feature enabled. The important values are entered as shown below.

- **Mode of Operation (MODE):** Route uses ISDN Signaling Link (ISLD)
- **D Channel number (DCH):** 1 -> this is D-Channel for SIP Trunk as defined in the Section 5.2
- **Interface Time 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 20 below.

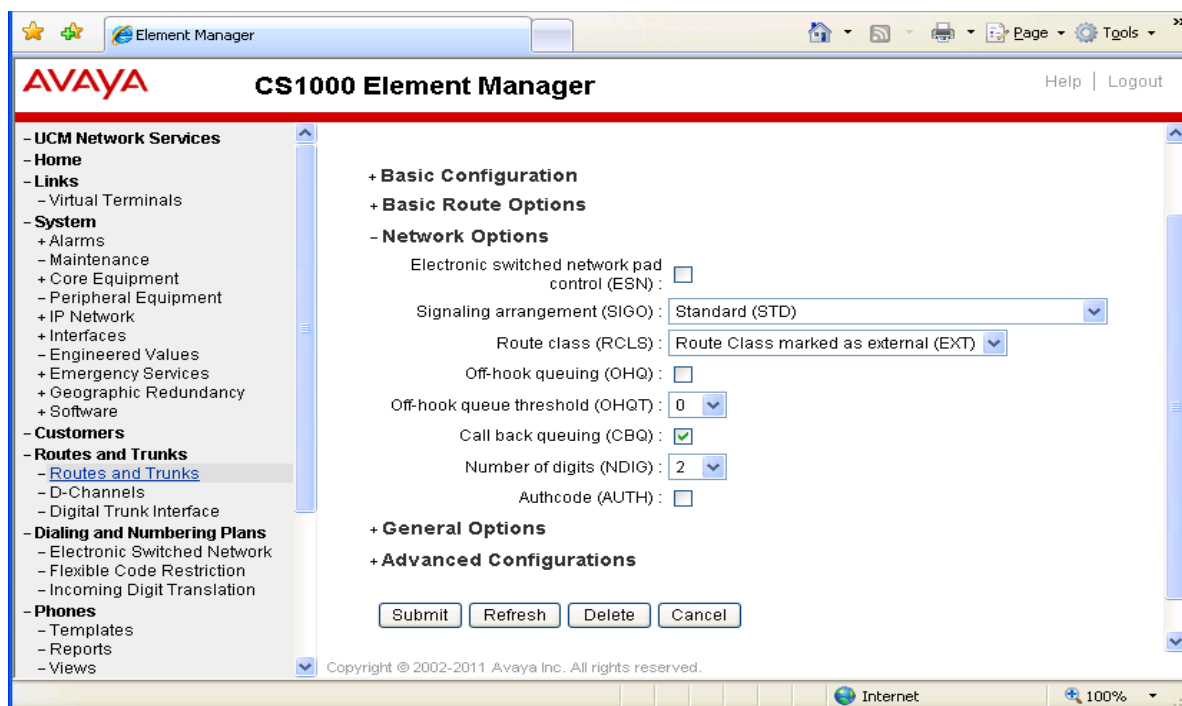
**Figure 20: Integrated services digital network option (ISDN) option page**

Continue expanding the **Basic Route Options** section and keep default values as shown in Figure 21.



**Figure 21: Basic Route Options of new Route**

Continue expanding the **Network Options** and keep default values as shown in Figure 22.



**Figure 22: Network Options of new Route**

Continue expanding the **General Options** section and keep default values as shown in Figure 23 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 (with sub-items: Alarms, Maintenance, Core Equipment, Peripheral Equipment, IP Network, Interfaces, Engineered Values, Emergency Services, Geographic Redundancy, Software), Customers, Routes and Trunks (with sub-items: Routes and Trunks, D-Channels, Digital Trunk Interface), Dialing and Numbering Plans (with sub-items: Electronic Switched Network, Flexible Code Restriction, Incoming Digit Translation), and Phones (with sub-items: Templates, Reports, Views). The 'Routes and Trunks' section is expanded, and 'Routes and Trunks' is selected. The main content area shows the 'General Options' section under 'Network Options'. The options are: 'M1 is the only controlling party on incoming calls (CPDC)' (checkbox), 'Dial tone on originating calls (DLTN)' (checkbox), 'Hold failure threshold (HOLD)' (text box with value '02 02 40'), 'Trunk access restriction group (TARG)' (text box with value '01'), 'Alternate trunk route for outgoing trunks (STEP)' (text box with value '(0 - 511)'), 'Actual outgoing toll digits to be ignored for code restriction (OABS)' (checkbox), '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 empty. 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

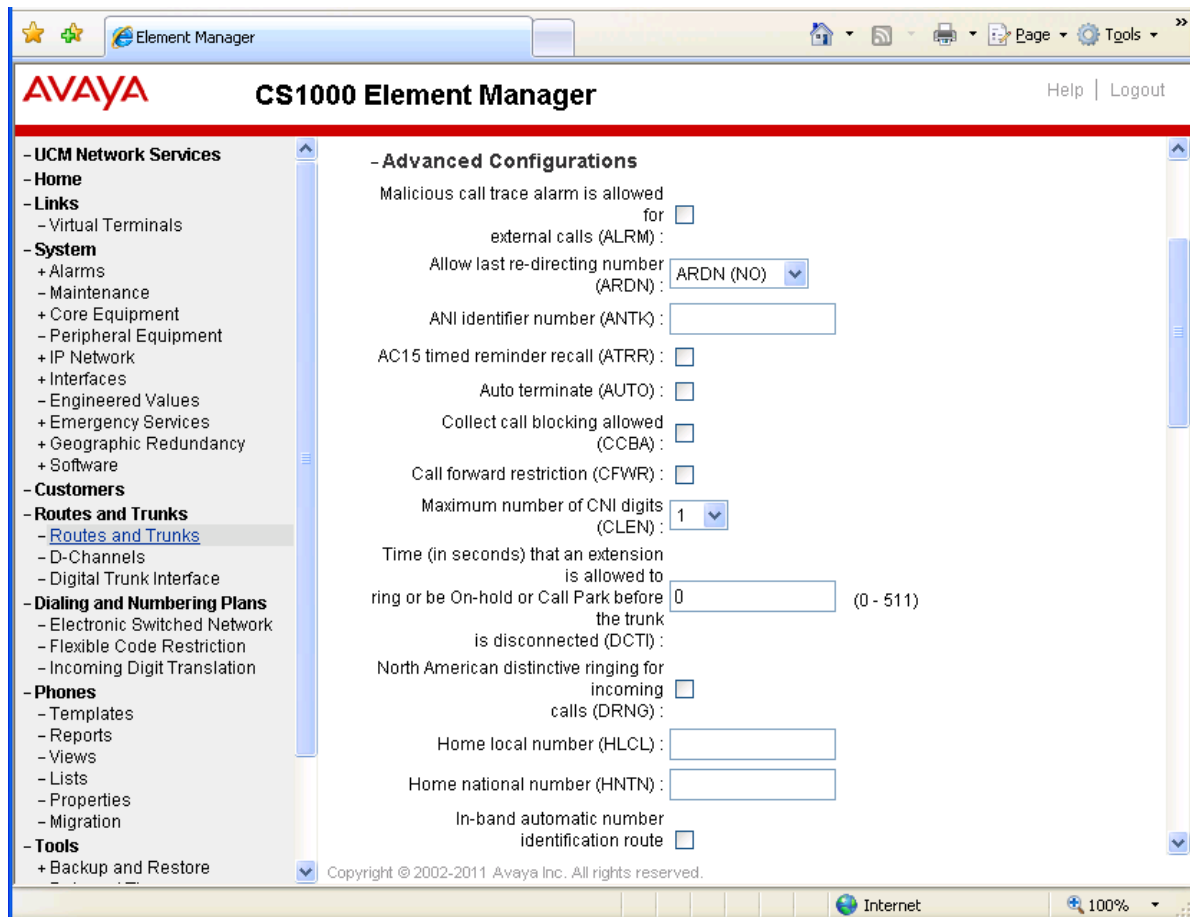
Submit Refresh Delete Cancel

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Internet 100%

**Figure 23: General Options of new Route**

Continue expanding the **Advanced Configurations** section and keep its values as shown in Figures 24, 25 and 26.



**Figure 24: 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

In-band automatic number identification route (IANI) : ☐  
 Incoming identifier send (ICIS) : ☒  
 Internal/external definition (IDEF) : Use network info (NET)   
 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) : 2   
 Protocol selection (PSEL) : DM-DM Protocol Selection (DMDM)   
 Preference trunk usage threshold (PTUT) : 0 (0 - 510)  
 Port type at far end (PTY) : Analog TIE trunks (ATT)

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**Figure 25: Advanced Configurations of new Route (cont)**

**AVAYA CS1000 Element Manager**

Help | Logout

Port type at far end (PTY) : 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 (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 (0 - 256)

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**Figure 26: Advanced Configurations of new Route (cont)**

## 5.5. Configure SIP Trunks

To configure SIP Trunks from the homepage of Element Manager, navigate to **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) which was created in the Section 5.4 and click on **Add trunk** button (not shown). The new Trunk page appears as shown in Figure 27.

In the **Basic Configuration** section, enter values as shown in Figure 27 below. Virtual trunks can be created as single or multiple by entering a number in the **Multiple trunk input number** field which is normally an increment of 32. For the **Member number** and **Channel ID for this trunk** field enter 1 if this is a first virtual trunk of this Route. This number is automatically incremented corresponding to the number of trunks created.

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 indicates the current location: 'Managing: 135.10.97.78 Username: admin' followed by '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', 'D-Channels', and 'Digital Trunk Interface'. 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.' and the browser status bar indicates 'Internet' and '100%' zoom.

Figure 27: Basic Configuration of new Trunk



Click on the **Edit** button of **Class of Service** field to enable necessary class of services of new trunks as shown in as shown in Figures 28 and 29 below. Click on the **Return Class of Service** button after completing this task.

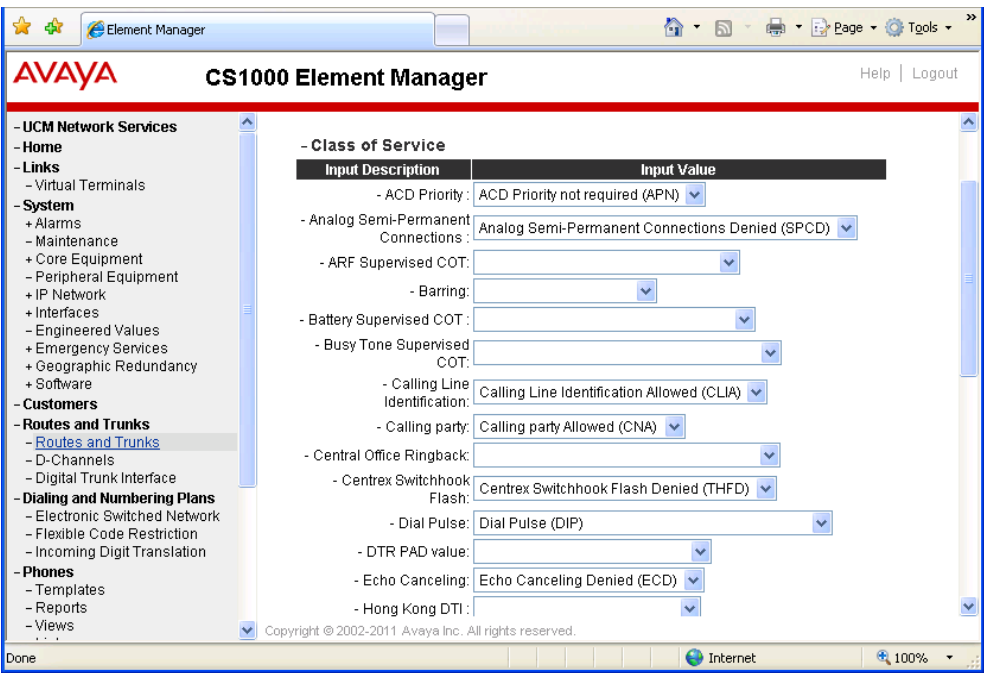


Figure 28: Class of Service of new Trunk

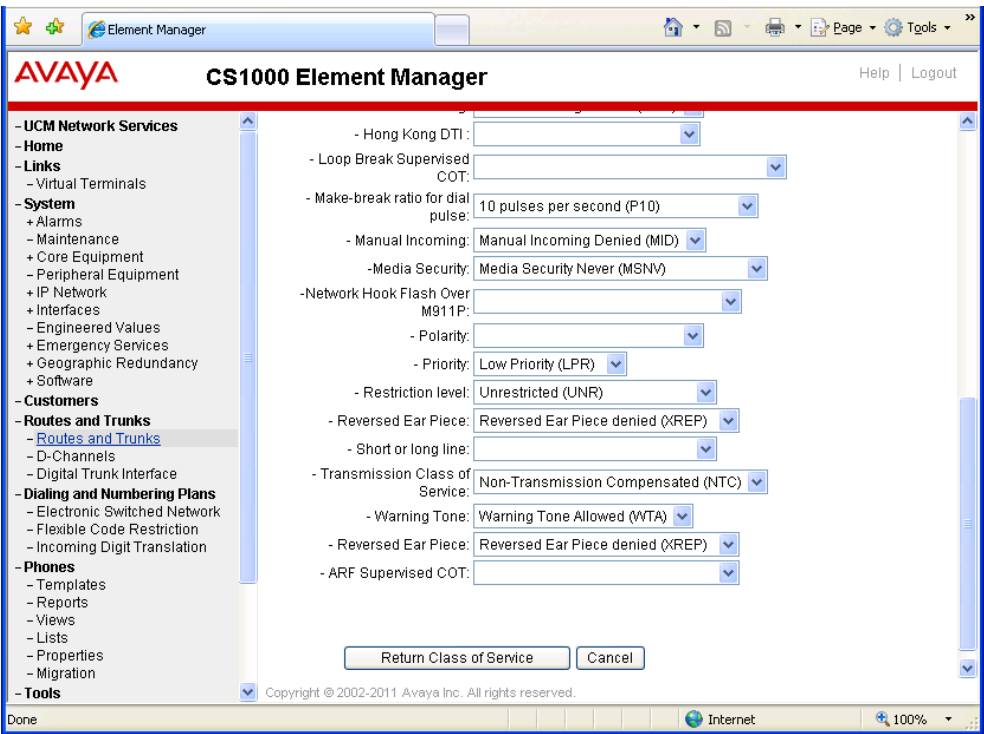


Figure 29: Class of Service of new Trunk (cont)

Continue expanding the **Advanced Trunk Configurations** section and keep its values as shown in Figure 30 below. Click on **Save** button 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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Internet 100%

**Figure 30: Advanced Trunk Configurations of new Trunk**

## 5.6. 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.

### 5.6.1. Configure Route List Index (RLI)

To configure Route List Index, from the home page of Element Manger, navigate to **Dialing and Numbering Plan > Electronic Switched Network**. The **Electronic Switched Network (ESN)** page appears as shown in Figure 31 below.

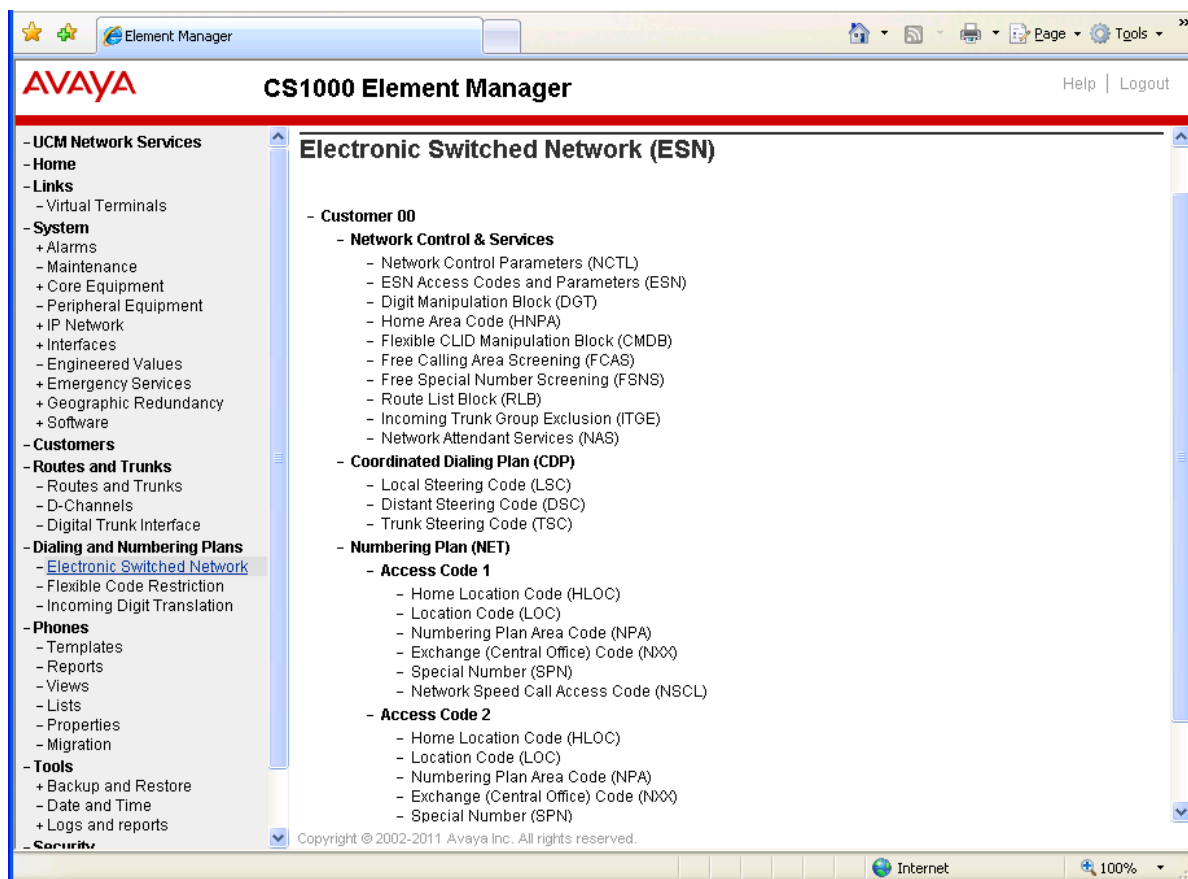
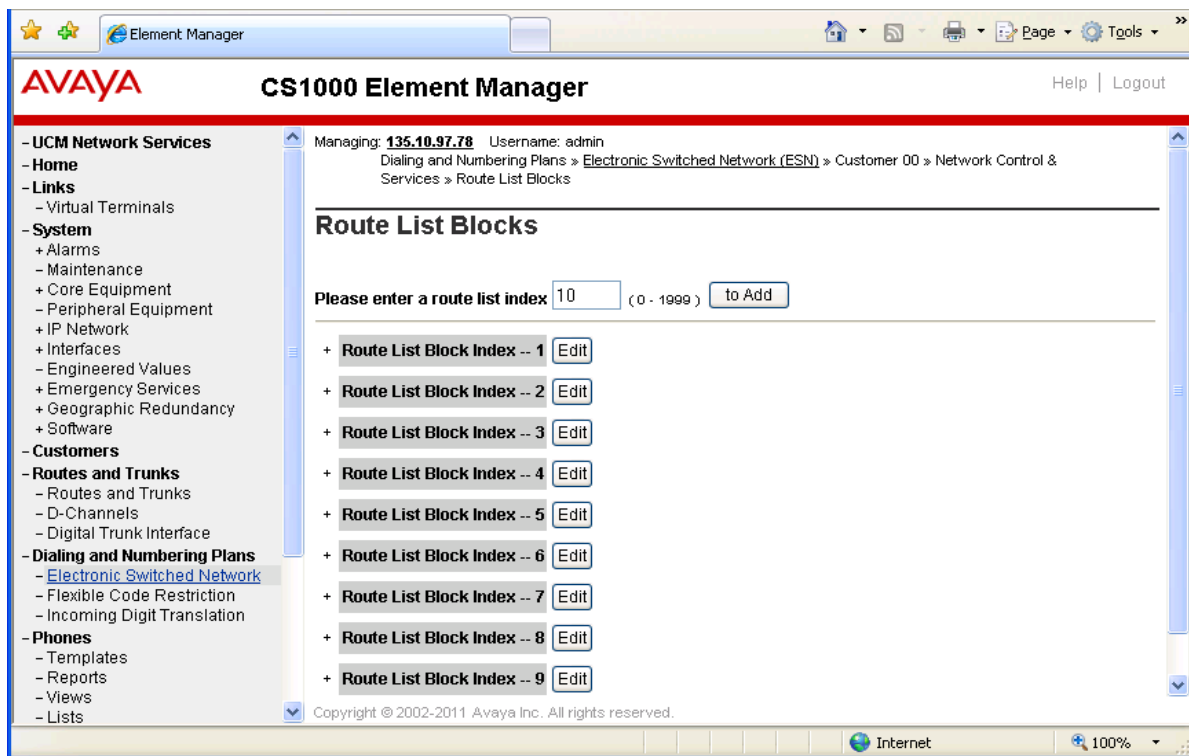


Figure 31: Electronic Switched Network (ESN) page

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 32. 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 32: Route List Blocks page**

The **General Properties** and **Indexes** sections of new route list index appear as shown in Figure 33 below. Keep all values at their defaults.

**AVAYA CS1000 Element Manager**

**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: ☐

Free Special Number Screening Index: ☐

Business Network Extension Route: ☐

Incoming CLID Table: 0 (0 - 100)

**Figure 33: General Properties and Indexes of Route List Block page**

The **Options** and **VNS Options** sections appear as shown in Figure 34. Select the **Route Number** in the dropdown list corresponding with the SIP route created in the Section 5.5. Keep all other values at their defaults.

Click **Submit** button to complete adding new route list index.

**AVAYA CS1000 Element Manager**

**Options**

Local Termination entry: ☐

Route Number: 1

Skip Conventional Signaling: ☐

Display Originator's Information: ☐

Use Tone Detector: ☐

Conversion to LDN: ☐

Expensive Route: ☐

Strategy on Congestion: No Reroute (NRR)

QSIG Alternate Routing Causes: QSIG Alternate Routing Cause 1

Preferred Routing: Preferred Route 1

ISDN Drop Back Busy: Drop Back Disabled (DBD)

ISDN Off-Hook Queuing Option: ☐

Off-Hook Queuing Allowed: ☐

Call Back Queuing Allowed: ☐

**VNS Options**

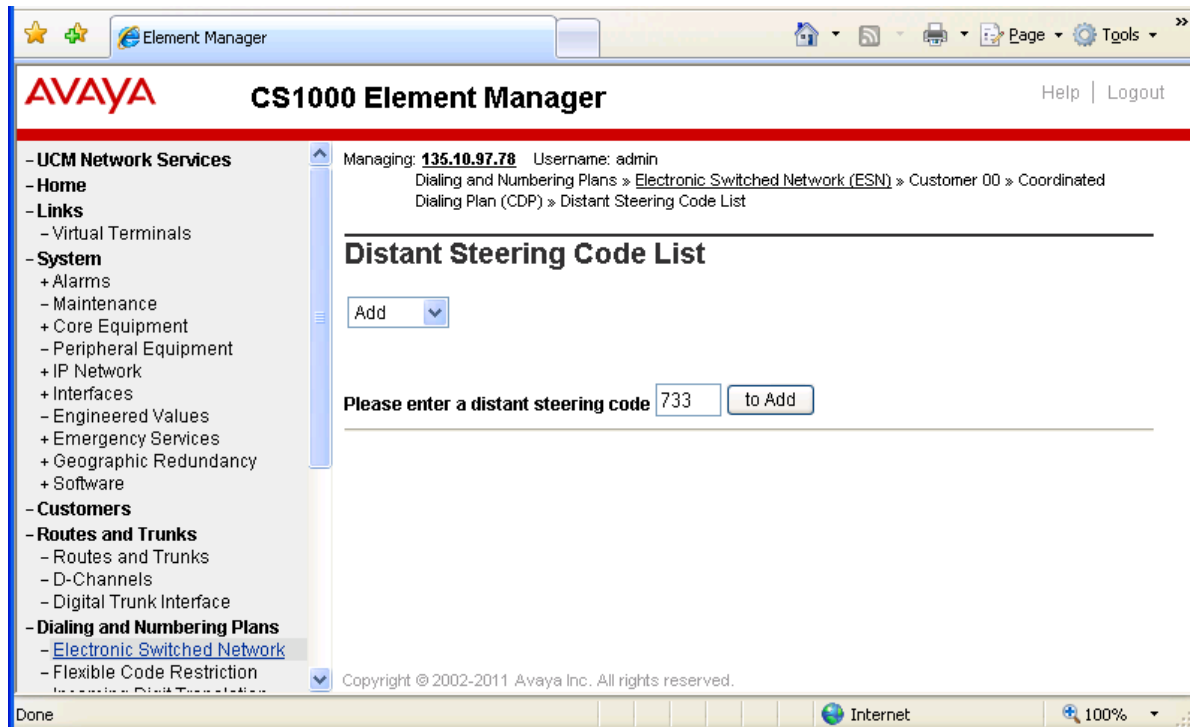
Entry is a VNS Route: ☐

Submit Cancel

**Figure 34: Options and VNS Options sections of Route List Blocks page (cont)**

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

To create 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 35 below. Select **Add** in the dropdown menu, enter the DSC code **733** in the field **Please enter a distant steering code** and then click on to **Add** button.



**Figure 35: Distant Steering Code List page**

The **Distant Steering Code** page appears as shown in Figure 36. Enter 5 in the field **Flexible Length number of digits**, because the length of dialed number to Movitas system is 5 digits., if 4 or 3 digits is planned, enter the corresponding length of digit in this field. 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.

The screenshot shows the 'Distant Steering Code' configuration page in the AVAYA CS1000 Element Manager. The left sidebar contains a navigation menu with categories like UCM Network Services, System, Customers, Routes and Trunks, and Dialing and Numbering Plans. The main content area has the following fields:

- Distant Steering Code: 733
- Flexible Length number of digits: 5 (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:

At the bottom right, there are 'Submit' and 'Cancel' buttons. The footer of the page includes the copyright notice: 'Copyright © 2002-2011 Avaya Inc. All rights reserved.'

**Figure 50: Distant Steering Code page**

## 5.7. Configure Phone Set and its Associated PCA

To configure phone set in the CS1000 system, log on to the Call Server as an administrator and use overlay LD 20. The following figure prints configuration of CS1000 phone set by using overlay 20; bold items in the Figures 51 and 52 need to be entered when creating a new phone set.

```
>ld 20

PT0000
REQ: prt
TYPE: 2050pc
TN 96 0 1 0
DATE
PAGE
DES
MODEL_NAME
EMULATED
KEM_RANGE

DES ROOM1
TN 096 0 01 00 VIRTUAL
TYPE 2050PC
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 00001
CUR_ZONE 00001
MRT
ERL 0
ECL 0
FDN
TGAR 1
LDN NO
NCOS 7
SGRP 0
RNPG 0
SCI 0
SSU
XLST
SCPW
```

**Figure 51: Sample of configuration of CS1000 Phone Set**



```

SFLT NO
CAC_MFC 0
CLS_CTD FBA WTA LPR MTD FNA HTA TDD HFA CRPD
MWD LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LND CNDA
CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBF
ICDD CDMF LLCN MCTD CLBD AUTU
GPUD DPUD DNDA CFXD ARHD CLTD ASCD
CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
UDI RCC HBTD AHD IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
DRDD EXRD
USMD USRD ULAD CCBF RTDD RBDD RBHD PGND FLXD FTTC DNDY DNO3 MCBN
FDSF NOVD VOLA VOUD CDMR PRED RECD MCDD T87A SBMD
KEM3 MSNV FRA PKCH MUTA MWTD DVLD CROD ELCD
CPND_LANG ENG
RCO 0
HUNT 54444
LHK 0
PLEV 02
PUID
UPWD
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 SCR 54006 0 MARP
CPND
CPND_LANG ROMAN
NAME ROOM 1
XPLN 13
DISPLAY_FMT FIRST, LAST

```

**Figure 52: Sample of configuration of CS1000 Phone Set**

To configure personal call arrangement (PCA) associated with a CS 1000 phone, use the same overlay as used to create phone set (LD 20). Please note that CS1000 phone can be digital, IP Unistim or SIP phones. The following Figures 53 and 54 print configuration of PCA by using overlay LD 20.

```
>ld 20

PT0000
REQ: prt
TYPE: pca
TN 96 0 1 1
DATE
PAGE
DES
MODEL_NAME
EMULATED

DES PCA
TN 096 0 01 01 VIRTUAL
TYPE PCA
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
MRT
ERL 0
ECL 0
FDN
TGAR 1
LDN NO
NCOS 7
SGRP 0
RNPG 0
SCI 0
SSU
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS CTD FBD WTA LPR MTD FND HTD TDD HFD CRPD
```

**Figure 53: Sample configuration of PCA**

```

CAC_MFC 0
CLS_CTD FBD WTA LPR MTD FND HTD TDD HFD CRPD
  MWD LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
  POD SLKD CCSD SWD LND CNDD
  CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
  ICDD CDMD LLCN MCTD CLBD AUTU
  GPUD DPUD DNDD CFXD ARHD CLTD ASCD
  CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
  UDI RCC HBTB AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
  DRDD EXR0
  USMD USRD ULAD CCBD RTDD RBDD RBHD PGND FLXD FTTC DNDY DNO3 MCBN
  FDSB NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD
  MSNV FRA PKCH MWTD DVLD CROD ELCD
CPND_LANG ENG
HUNT
PLEV 02
PUID
UPWD
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY 00 SCR 54006 0
  CPND
    CPND_LANG ROMAN
    NAME ROOM 1
    XPLN 13
    DISPLAY_FMT FIRST, LAST
01 HOT P 5 73301
02
03

```

**Figure 54: Sample of configuration of PCA**

In the PCA configuration, KEY 01 HOT P <DN> is critical in routing the call to Movitas MvPBX system. Whenever a call comes in to a CS1000 phone set associated with PCA, the call is also routed to the Movitas SIP user which has a respective DN in each PCA configuration based on this field. In this sample DN 73301 is assigned in KEY 1 HOT P to route call to Movitas MvPBX.

**Figure 55** shows directory number (DN) **54006** in the CS 1000 system are mapped to one real phone 2050PC and one PCA, as described above the PCA is responsible to route call to Movitas system.

```
>ld 20

PT0000
REQ: prt
TYPE: dn
TYPE DNB
CUST 0
DN 54006
DATE
PAGE
DES

DN 54006
  CPND
    CPND_LANG ROMAN
    NAME ROOM 1
    XPLN 13
    DISPLAY_FMT FIRST,LAST
  TYPE SL1
  TN 096 0 01 01 V KEY 00    DES RIMPCA  28 AUG 2011
    (PCA )
  TN 096 0 01 00 V KEY 00 H MARP DES ROOM1   28 AUG 2011
    (2050PC)
```

**Figure 55: Sample of one DN associated with phone set and PCA**

## **6. Configure Movitas MvPBXSystem**

This document assumes that the Movitas MvPBX system was properly installed and configured by a Movitas engineer. This section provides the steps of how to configure Movitas MvPBX with an Avaya Communication Server 1000 system.

The following summarizes the tasks which need to be done on the Movitas MvPBX and Dreams Cancun application:

- Configure Business account.
- Enable the Business account for calling
- Configure SIP Trunk in MvPBX
- Assign SIP Trunk to Business Account
- Setup Business Extensions
- Setup Call Destinations for the Extension
- Deploy Applications Via App Stores
- User Creates Account on Application
- User Creates Account on Application
- Assign User to Room Phone From Application

### **6.1. Configure Business Account**

As a partner on Movitas.com, a business account is established by signing up, choosing a name for the account and providing any necessary billing information. Once this is established the content, users, designs and business extensions can be managed via the Movitas interface.

### **6.2. Enable the Business Account for Calling**

After a business account is established, a Movitas admin can set up the business to enable calling functionality on the account. This will result in an additional account setting tab where the SIP trunk can be set, IP addresses can be setup to restrict calls to certain sites, push notifications can be configured and more. For the CS1000 testing, the only required field is the SIP Trunk, which will be assigned in step 6.3 below.

Test Dreams Account Settings
« back to the Main Menu

Basic Business Info
Manage Locations
Billing & Payment
Upgrade!
Business Price Profiles
Connect to Networks
Manage Call Settings

### Account Settings

#### Calling Settings

☐ Long Distance Enabled
☐ Local Calls Enabled
☐ International Warning
☐ Bridge Numbers Enabled
☐ Extensions Enabled
☐ Restrict Outgoing SIP Calls to Designated IP's
☐ Restrict Incoming SIP Calls to Designated IP's
☐ Restrict Outgoing External Calls to Designated IP's

Restricted IP's (separate multiple ID's by commas and subnets are supported)

Primary SIP Trunk Extension

Allowed Country Codes

Secondary SIP Trunk Extension

Android App URL

iOS App URL

Consumer Caller ID Number Format

Consumer Caller ID Name Format

Extension Caller ID Number Format

Extension Caller ID Name Format

#### iPhone App Settings

Upload Certificate (.p12)

p12 File Password (optional):

Confirm Password:

#### Android App Settings

Account Type:

Sender ID:

Password:

Confirm Password:

Application ID:

**Figure 56: Dreams Account Settings page**

### 6.3. Configure SIP Trunk in MvPBX

Once a business is setup, Movitas can assign a SIP Trunk username and password for the Avaya system. The required settings for this are:

- Name: a short name to use as an identifier for the trunk
- Fullname: a human-friendly used to describe the trunk
- Secret: the password used for authentication from the CS1000
- Transport: should be set to UDP
- Directmedia: should be set to nonat
- Default county code: the numerical country code for outgoing international call parsing
- PSTN prefix: prefix for outbound calls

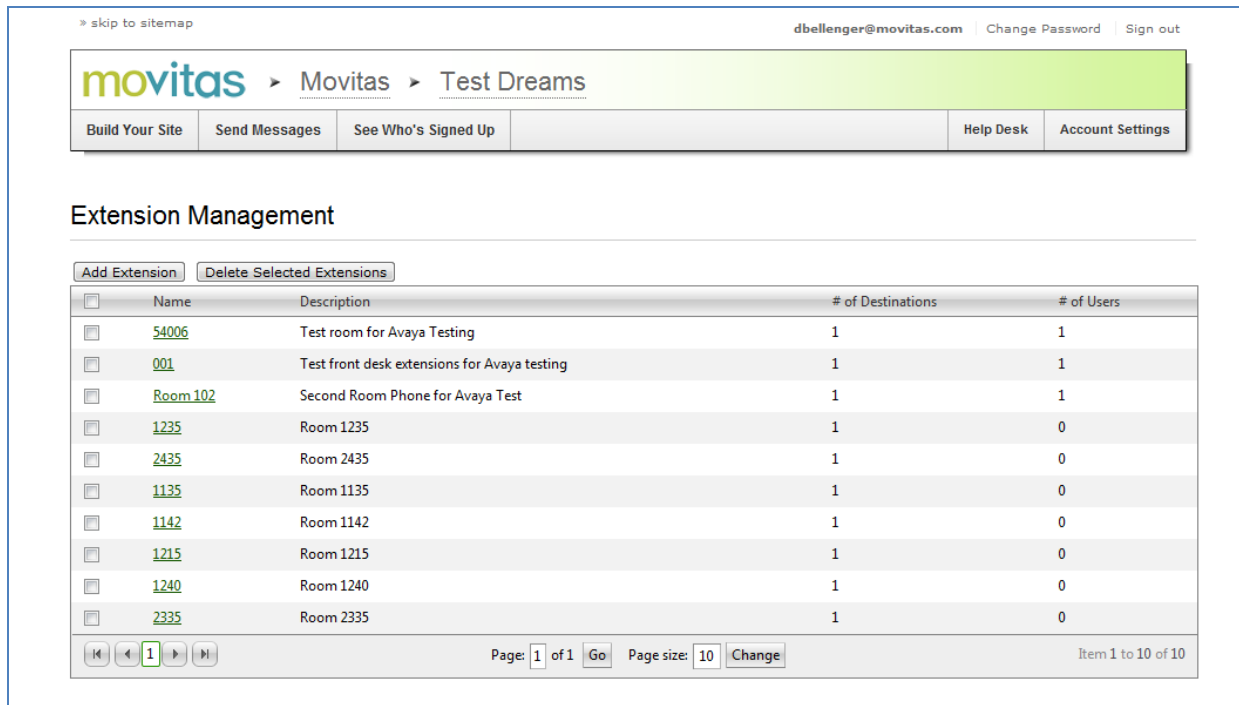
Once established, the credentials are provided to complete the setup on the CS1000 in Section 5.1.

### 6.4. Assign SIP Trunk to Business Account

Following the trunk creation, the account administrator can return to the business account for the application and enter the SIP Trunk name into the Primary SIP Trunk field.

## 6.5. Setup Business Extensions

Once the trunk is established, administrators can navigate to the room extensions page where they can create and manage extensions.



The screenshot displays the 'Extension Management' page in the Movitas application. At the top, there is a navigation bar with the Movitas logo and breadcrumb links: 'Movitas > Test Dreams'. Below this is a menu bar with options: 'Build Your Site', 'Send Messages', 'See Who's Signed Up', 'Help Desk', and 'Account Settings'. The main content area is titled 'Extension Management' and includes two buttons: 'Add Extension' and 'Delete Selected Extensions'. A table lists the following extensions:

<input type="checkbox"/>	Name	Description	# of Destinations	# of Users
<input type="checkbox"/>	<a href="#">54006</a>	Test room for Avaya Testing	1	1
<input type="checkbox"/>	<a href="#">001</a>	Test front desk extensions for Avaya testing	1	1
<input type="checkbox"/>	<a href="#">Room 102</a>	Second Room Phone for Avaya Test	1	1
<input type="checkbox"/>	<a href="#">1235</a>	Room 1235	1	0
<input type="checkbox"/>	<a href="#">2435</a>	Room 2435	1	0
<input type="checkbox"/>	<a href="#">1135</a>	Room 1135	1	0
<input type="checkbox"/>	<a href="#">1142</a>	Room 1142	1	0
<input type="checkbox"/>	<a href="#">1215</a>	Room 1215	1	0
<input type="checkbox"/>	<a href="#">1240</a>	Room 1240	1	0
<input type="checkbox"/>	<a href="#">2335</a>	Room 2335	1	0

At the bottom of the table, there is a pagination control showing 'Page: 1 of 1', a 'Go' button, 'Page size: 10', a 'Change' button, and 'Item 1 to 10 of 10'.

**Figure 57: Setup Business Extensions**

Click “Add Extension” to be taken to the new extension page as shown below:

The screenshot shows a web interface for setting up a business extension. At the top, there's a header with the Movitas logo and navigation links: "Build Your Site", "Send Messages", "See Who's Signed Up", "Help Desk", and "Account Settings". The user is logged in as "dbellenger@movitas.com". The main content area is titled "Room Extension" and includes a "back to Manage Extensions" link. The "Extension Details" section contains fields for "Extension Name", "Description", "Caller ID", and "Third Party Extension", along with a checkbox for "Allow Voicemails". The "SIP Login" section shows "SIP Username" and "SIP Password" fields, both with the text "Username Generated On Save" and "Password Generated On Save" respectively. A "Save Extension" button is at the bottom.

**Figure 58: Setup Business Extensions (cont)**

Enter the following information:

- Extension Name – a familiar name that can be used to identify the extension (ex. Room 101)
- Description – a description of the name if required
- Caller ID – if a different caller ID is required for the extension it can be entered here. (Note: caller ID and caller Number for consumer extensions and business extensions can also be formatted on the call settings page using variables such as room number, first name, last name, booking last name, caller ID, extension name, etc).
- Third Party Extension – for Avaya implementations, the PCA number should be added here in order to map calls to these extensions. This number must be unique for each business extension



Once all the settings are complete, click Save Extension and the extension will now be enabled and call destinations can be configured such as users, DID's, SIP extensions, or CS1K extensions. An example of the updated page is below.

**Room Extension** \* back to Manage Extensions

---

**Extension Details**

Extension Name:

Description:

Caller ID:

Third Party Extension:

☐ Allow Voicemails

**Manage Extension Users**

Add or remove users from this extension number. Click on a user's name to view and edit their information.

[Add User to Extension](#)

User's Name	User's Email	
Doug Bellenger	doug.bellenger@gmail.com	<a href="#">Delete</a>

1 Page 1 of 1 items 1 to 1 of 1

---

**SIP Login**

SIP Username:

SIP Password:

[Save Extension](#)

---

**Extension Destinations**

Calls made to this extension will be directed to:

☒ one destination number at a time, in the order that they are listed. (Drag and drop to re-order)

☐ all destination numbers at the same time.

[Add Destination](#)

Priority	Name	Destination #	Type	
1	54006 Avaya Extension	54006	Extension	<a href="#">Edit</a> <a href="#">Delete</a>

1 Page 1 of 1 items 1 to 1 of 1

**Figure 59: Setup Business Extensions (cont)**

## 6.6. Setup Call Destinations for the Extension

In order to forward calls from a user to the room, in addition to setting up the Third Party Extension with the PCA number, the DN also should be added as an extension as shown above. (Warning: setting up the PCA as a call destination is not recommended due to the creation of a loop between the two systems). In addition to hotel extensions, a call destination can be pointed at a SIP address, a DID, or a user in the system.

## 6.7. Deploy Applications Via App Stores

In order to complete the setup of the applications, Movitas will package the content and calling functionality into the required apps and enter them into the appropriate app store. Once completed the application will be available for download and the content and extensions can be updated via the Movitas portal. Any changes to the SIP Trunk for the business, IP addresses or other settings can be updated as well.

## 6.8. User Creates Account on Application

In order to place calls via the application following the setup, a user will sign up for an account, log in to the application and then they will be registered with MvPBX automatically. Once they are logged in, they can navigate to the dial pad where they can dial an extension direct or dial another user account.

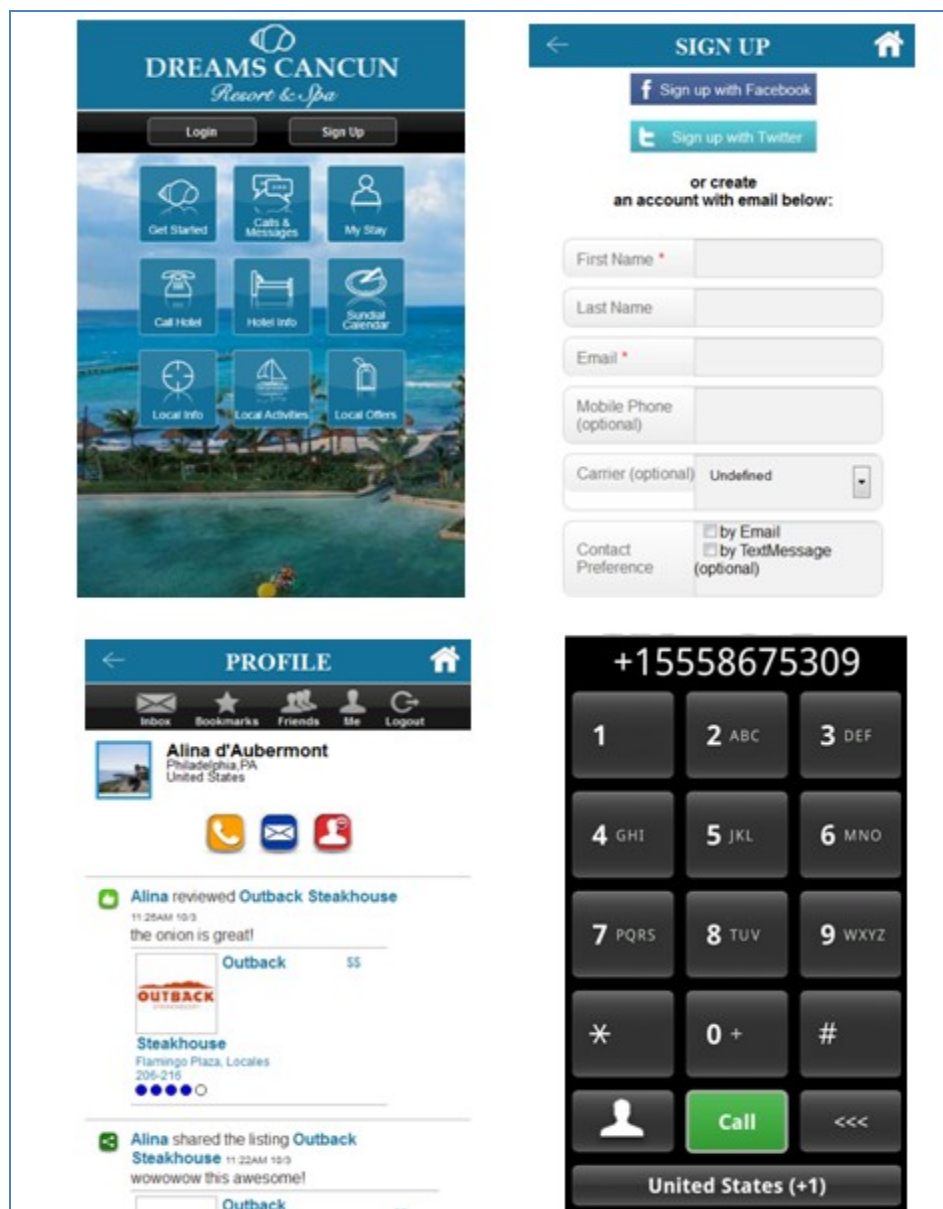


Figure 60: Sample of User create account on the application

## 6.9. Assign User Such as Front Desk to Extension

Once a user account is setup, administrators can assign users to extensions by accessing the business extension on the Movitas administrator screen and adding the email address of the user on the right side of the page. A user can also be deleted from this page by clicking the “delete” by their name once they’re added.

**Room Extension**

[\\* back to Manage Extensions](#)

**Extension Details**

Extension Name:

Description:

Caller ID:

Third Party Extension:

☐ Allow Voicemails

**SIP Login**

SIP Username: r-3

**Manage Extension Users**

Add or remove users from this extension number. Click on a user's name to view and edit their information.

[Add User to Extension](#)

User's Name	User's Email	
Doug Bellenger	doug.bellenger@gmail.com	<a href="#">Delete</a>

1 Page 1 of 1, Items 1 to 1 of 1

Figure 61: Assign Front Desk Extension to User

## 6.10. Assigning User to Room Phone From Application

Pending the availability of web services into the PMS, a page may be setup to enable forwarding of the room phone as outlined below. This will associate the user with the appropriate extension that calls can be forwarded or received from. This can also be handled by web services.

[Inbox](#) [Bookmarks](#) [Friends](#) [Me](#) [Logout](#)

**My Room Info**

Please fill in your last name and room number to turn on My Stay functions.

Last Name

Room No.

[Submit](#)

**My Bill**

Click the button to view your current charges.

[View Bill](#)

**My Room Phone**

Click the button to activate call forwarding from your room phone to this phone.

[Enable Room Forwarding](#)

**Figure 62: Assign Room Extension to User**

## **7. Verification Steps**

The following are typical steps to verify the interoperability between the Movitas system and Avaya Communication Server 1000 Release 7.5.

- Place a call to Front Desk phone which is one of three CS1000 extensions associated with Movitas SIP users
- The Front Desk phone in the CS1000 system rings and the Movitas SIP user associated with the Front Desk phone also rings.
- Accept the call on the Movitas SIP user, the Front Desk phone in the CS1000 stops ringing and become idle.
- Check two-way audio path between the caller and Movitas SIP user.

## **8. Conclusions**

All of the executed test cases have passed and met the objectives as outlined in **Section 2**. The Movitas SIP MvPXB system and its respective iPhone, Android and Blackberry applications considered compliant with Avaya Communication Server 1000 Release 7.5.

## **9. Additional References**

Product documentation for Avaya products may be found at:

<https://support.avaya.com/css/Products/>

Product documentation for Movitas' product may be found at:

<http://www.movitas.com>

### **Avaya Communication Server 1000 Documents:**

-Avaya Communication Installation and Commissioning, Doc# NN43041-310, Issue 05.04, Date May 2011.

-Avaya Communication Server 1000 Unified Communications Management Common Services Fundamentals, Doc # NN43001-116, Issue 05.11, Date June 2011.

-Avaya Communication Server 1000 Co-resident Call Server and Signaling Server Fundamentals, Doc # NN43001-509, Issue 03.02, Date June 2011.

-Avaya Communication Server 1000 Element Manager System Reference - Administration, Doc# NN43001-632, Issue 05.09, Date July 2011.

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