Avaya Solution and Interoperability Test Lab

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.

Telephone: 888-343-3721 ext. 1
Email: 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.

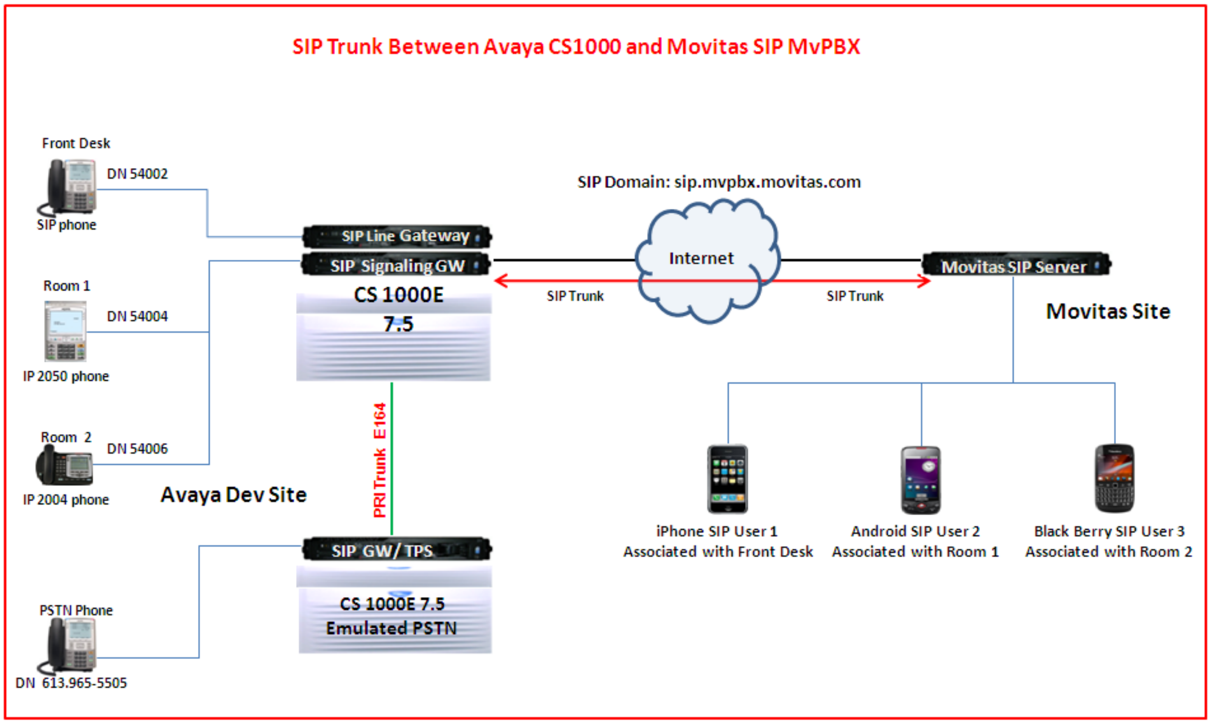

Figure 1: Network Diagram Configuration

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.

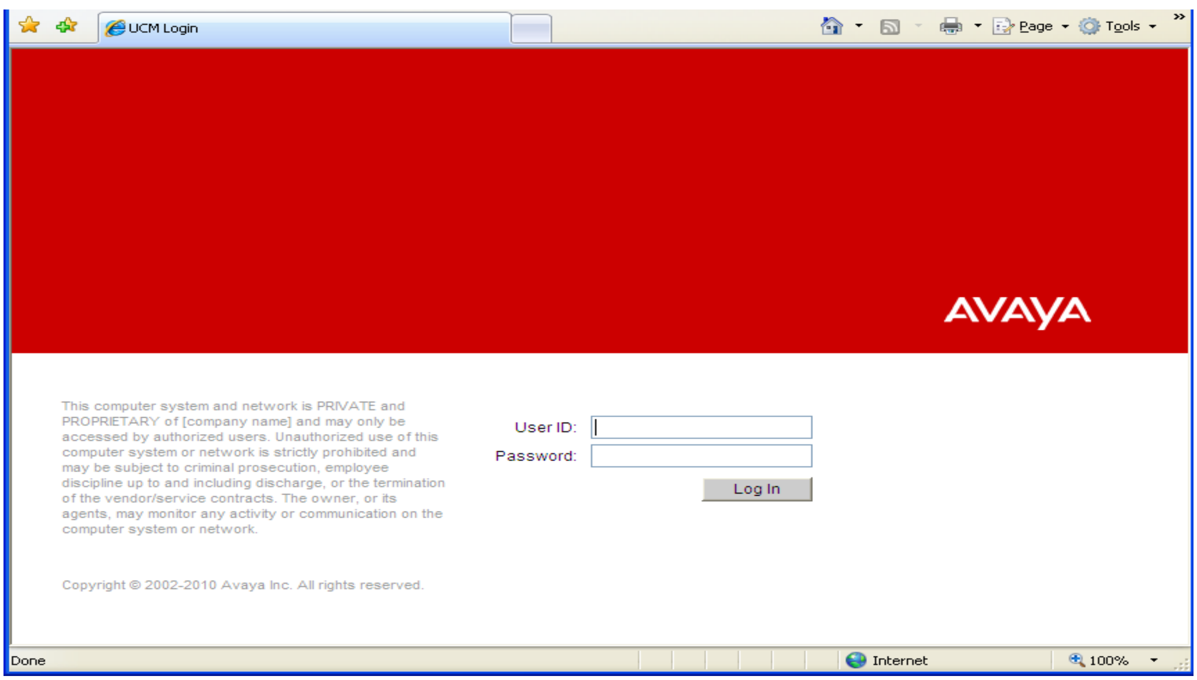

Figure 2: UCM Login window

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.

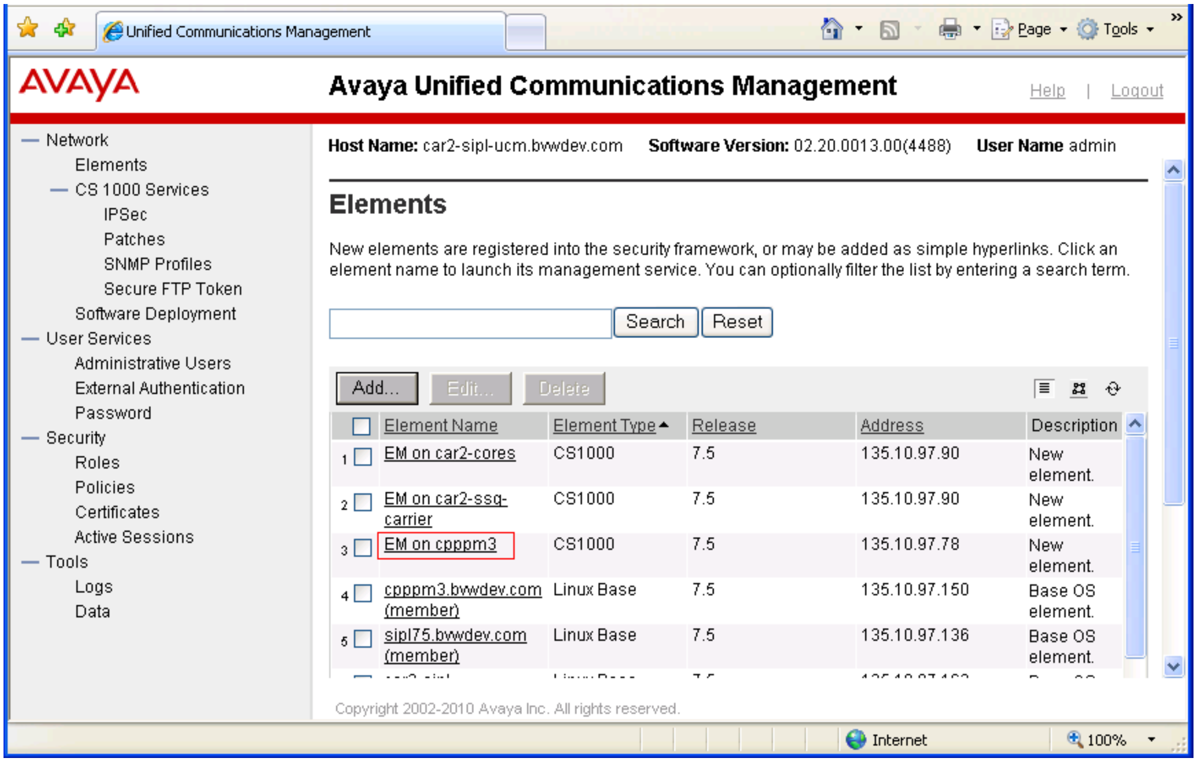

Figure 3: UCM Home Page

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.

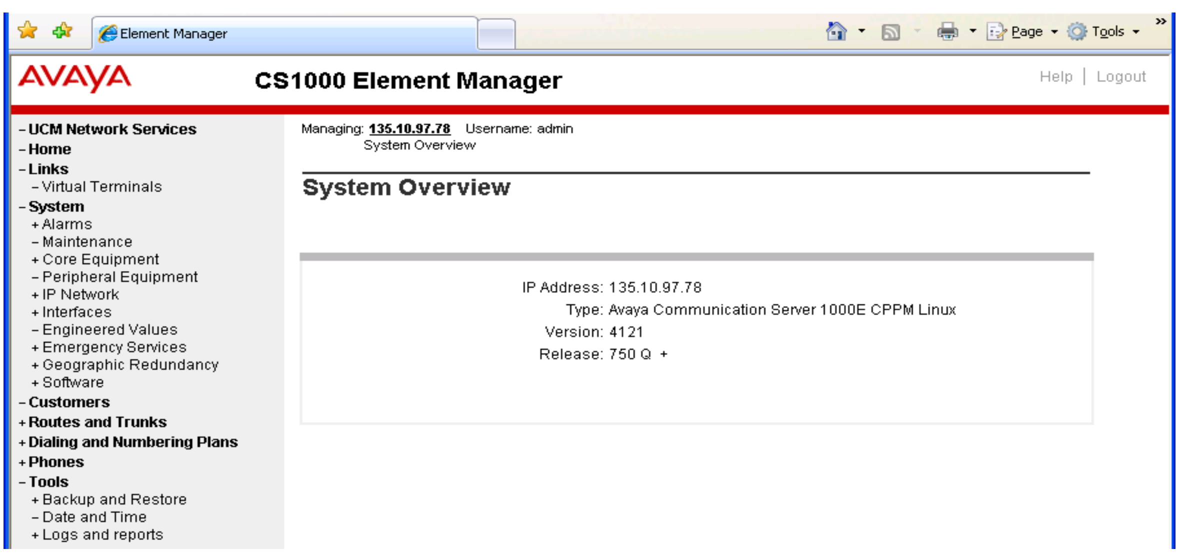

Figure 4: CS1000 Element Manager Home page

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.

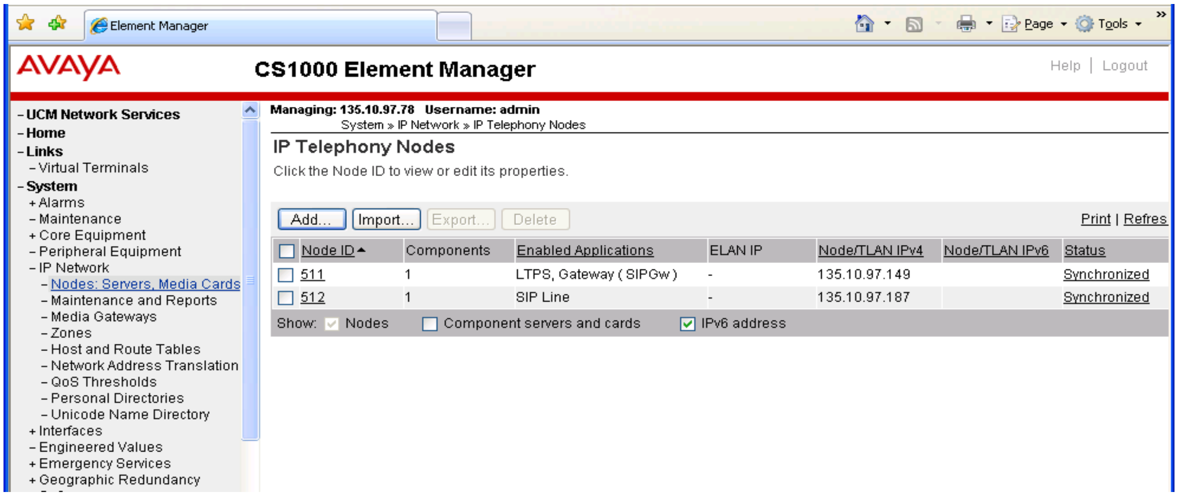

Figure 5: IP Telephony Nodes Page

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.

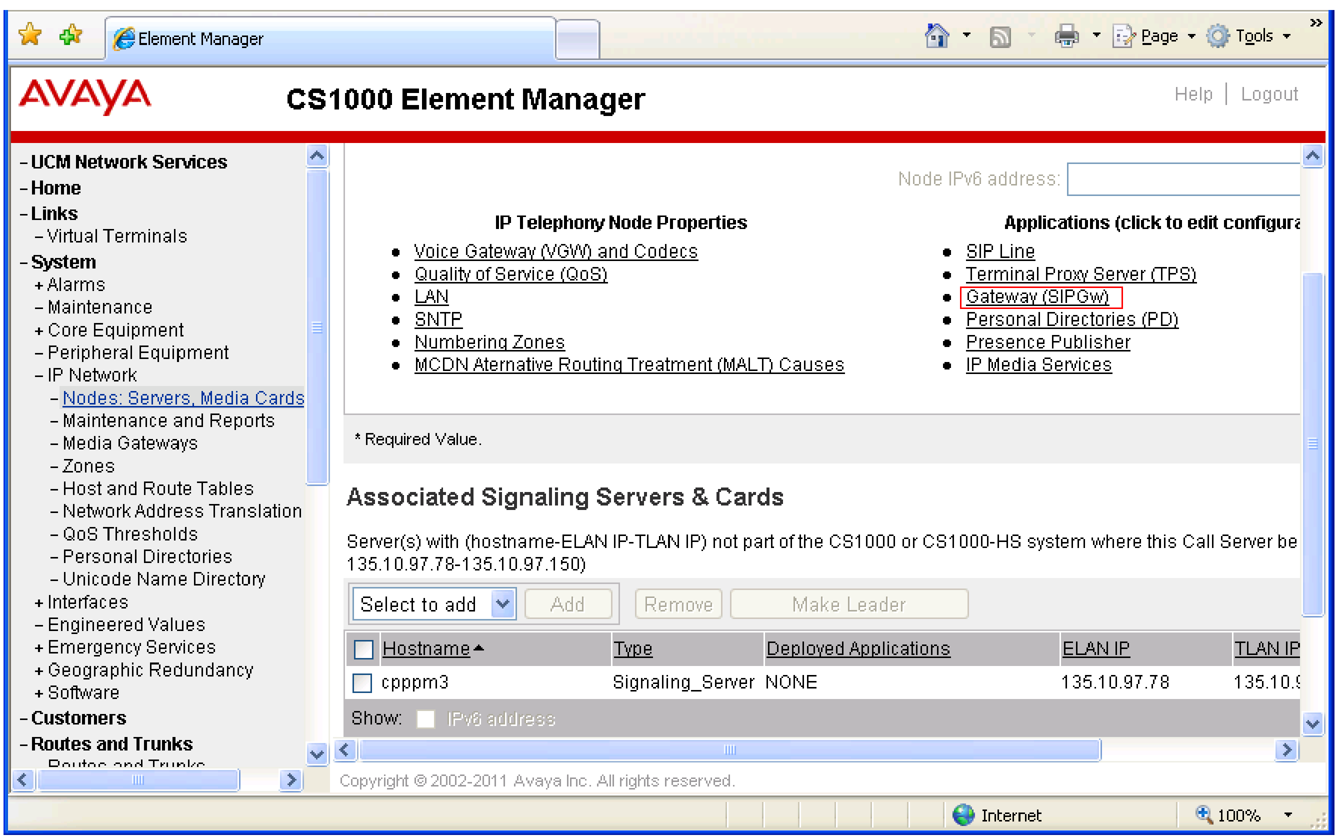

Figure 6: IP Telephony Node Detail Page

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.mvpbx.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.

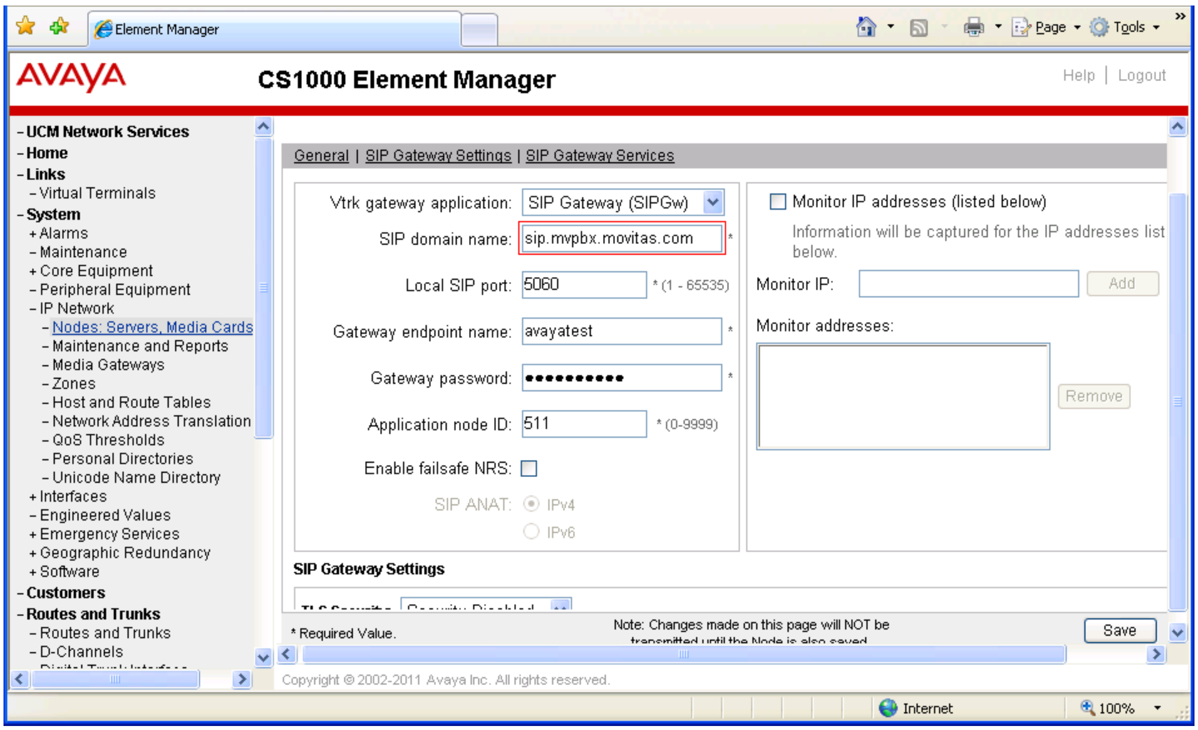

Figure 7: Node ID: 511 – Virtual Trunk Gateway Configuration Details

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.

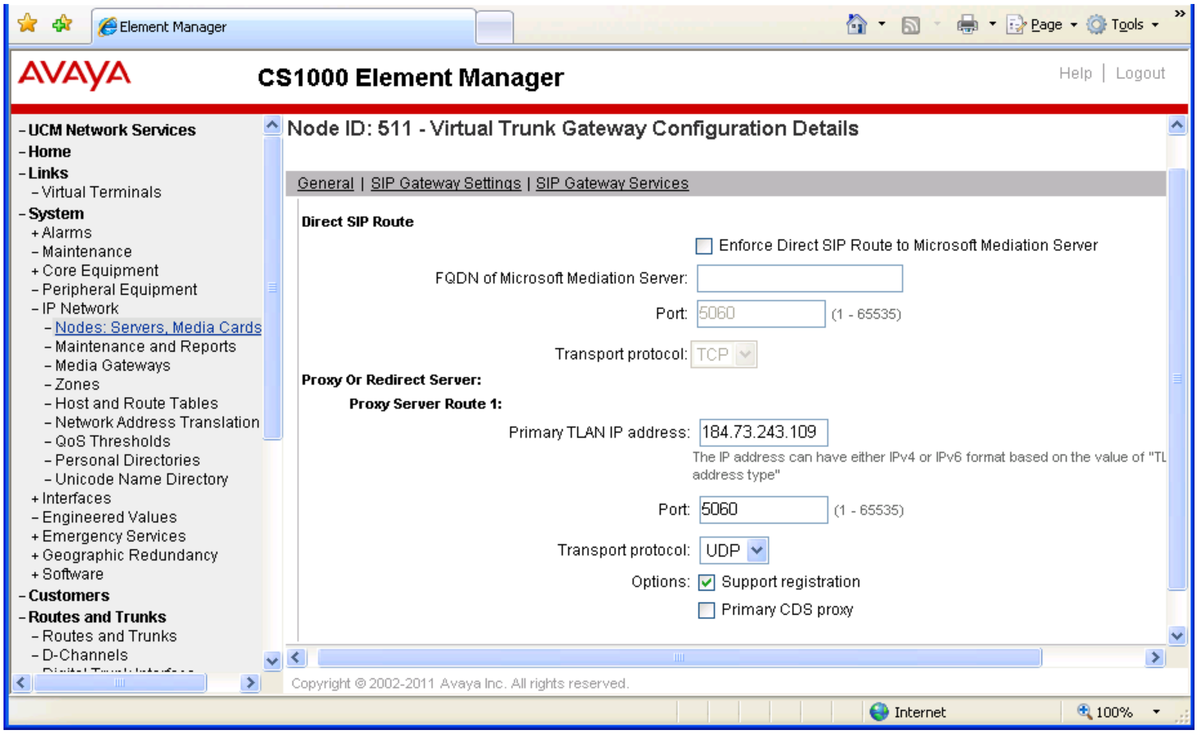

Figure 8: The SIP Gateway Setting of Telephony Node

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.

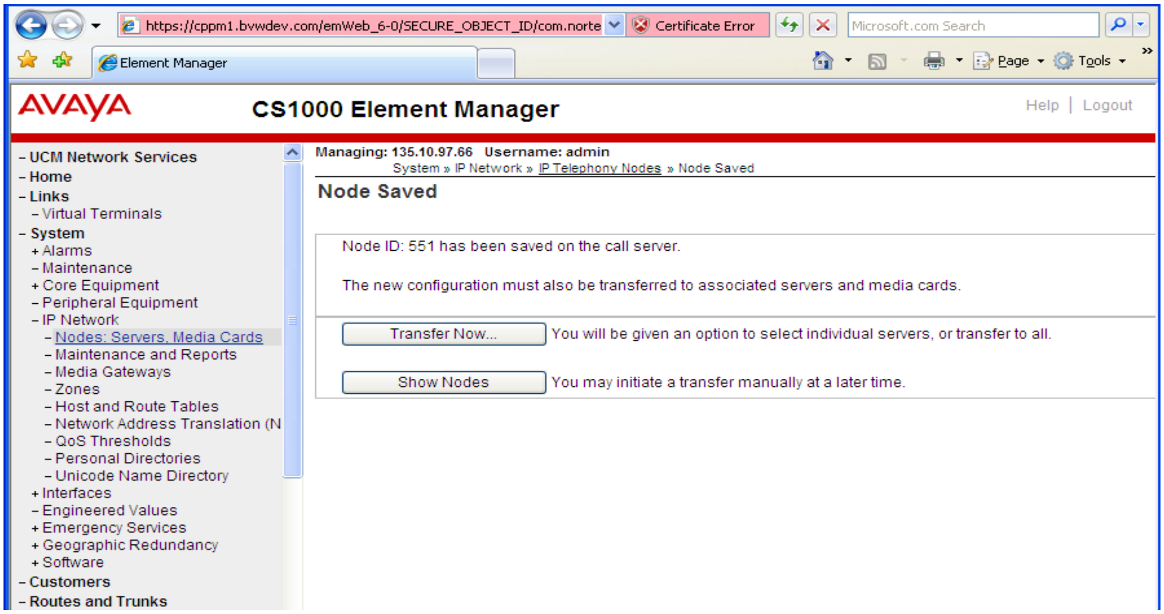

Figure 9: Node Saved Page

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.

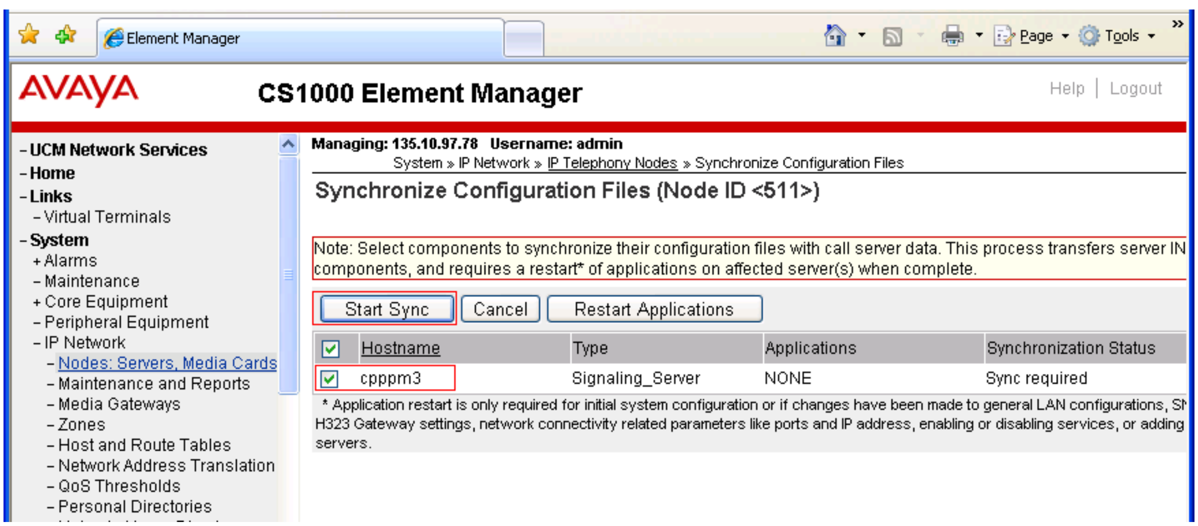

Figure 10: Synchronize Configuration Files (Node ID <551>) page

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.

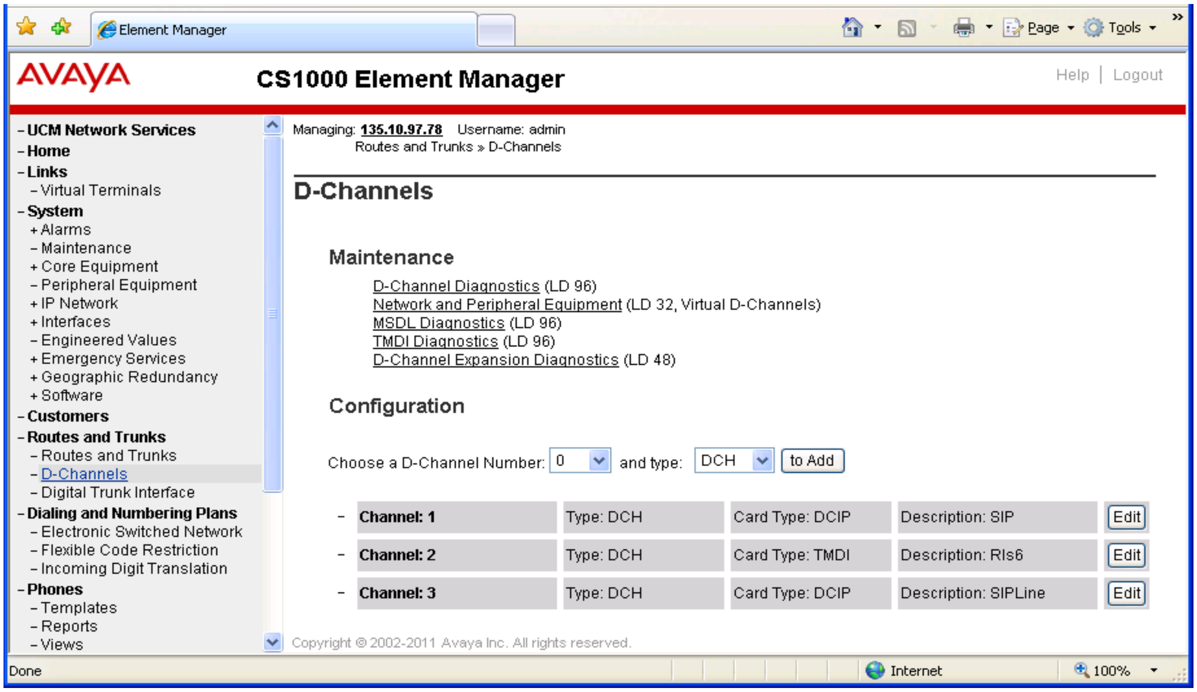

Figure 11: D-Channels page

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 to **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.

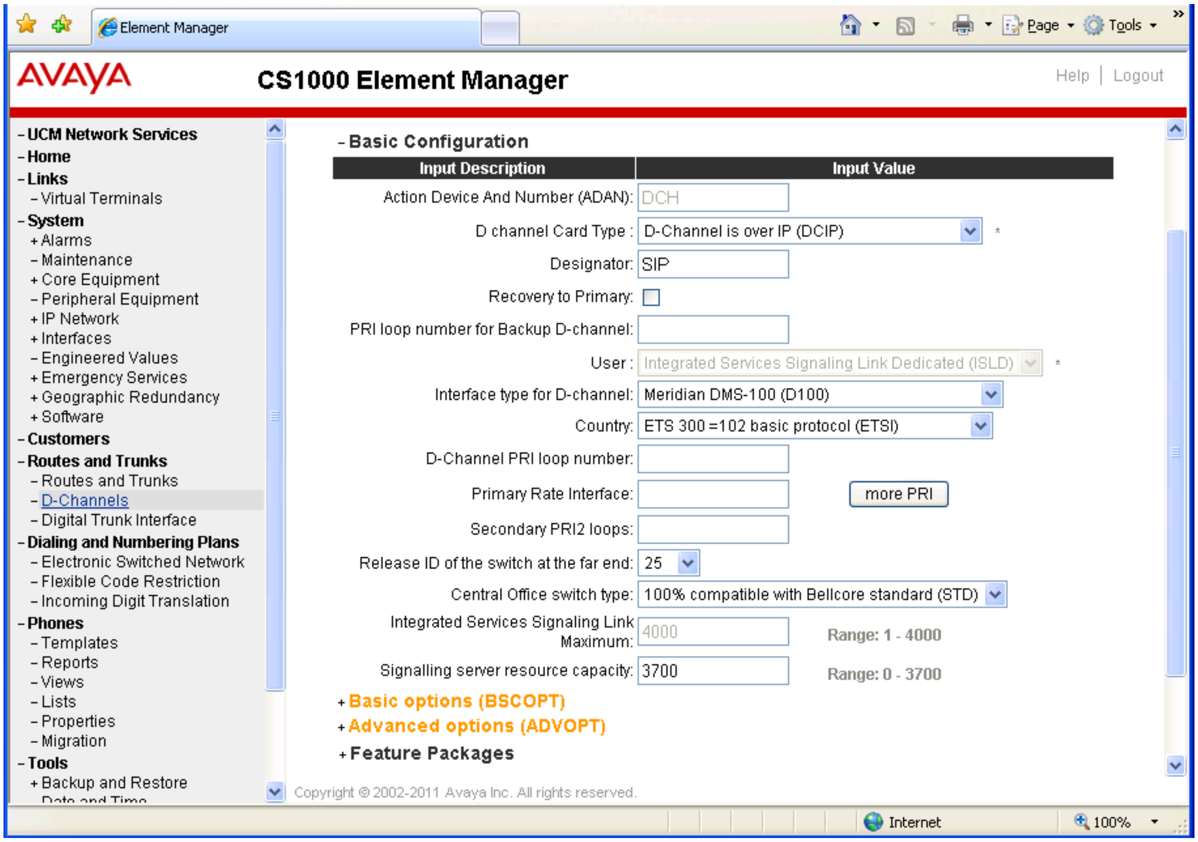

Figure 12: Basic Configuration section of D-Channel

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.

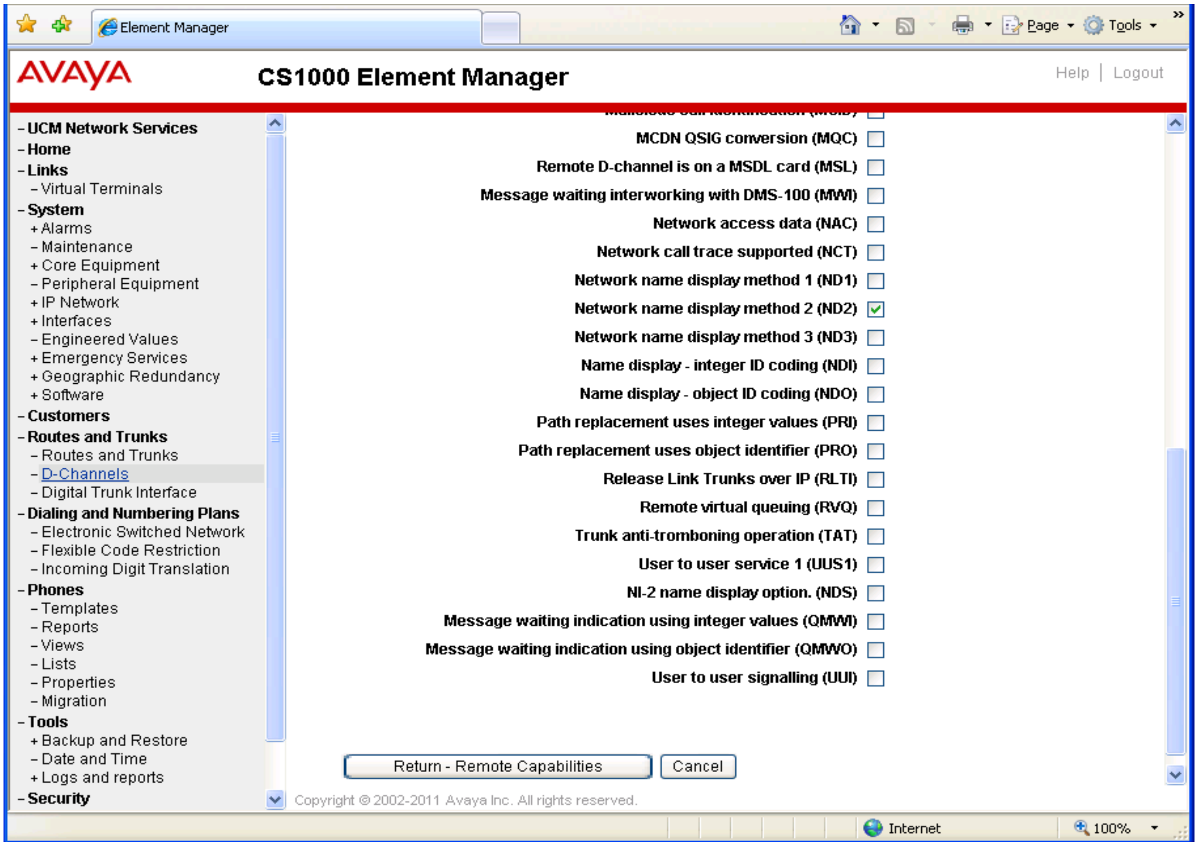

Figure 13: Remote Capability page

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.

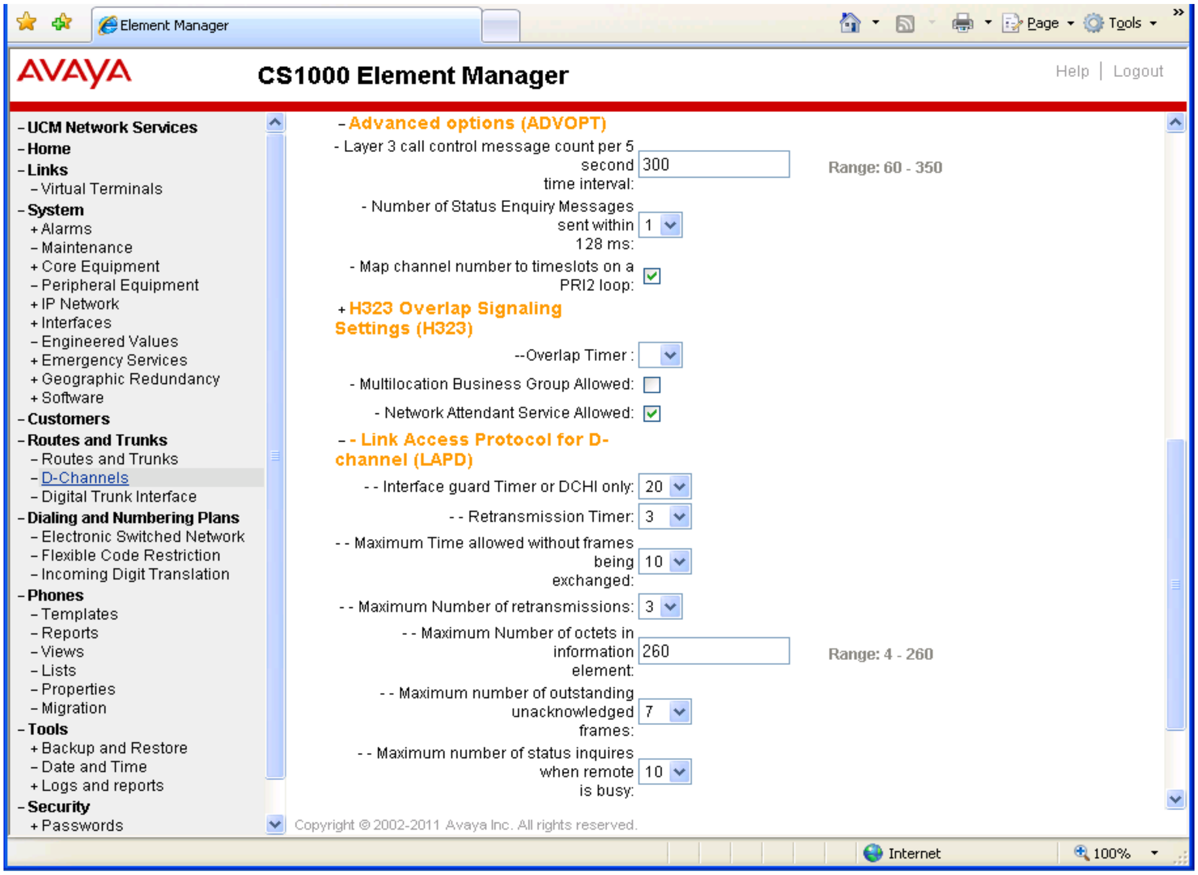

Figure 14: Advanced options (ADVOPT)

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.

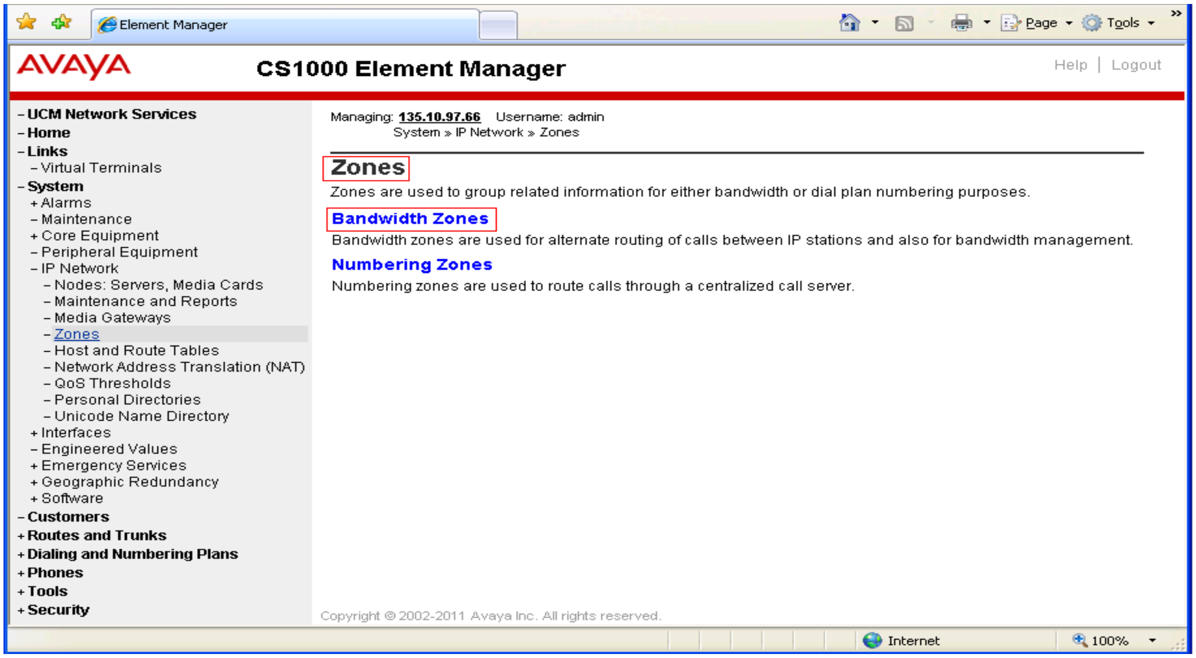

Figure 15: Zones Page

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.

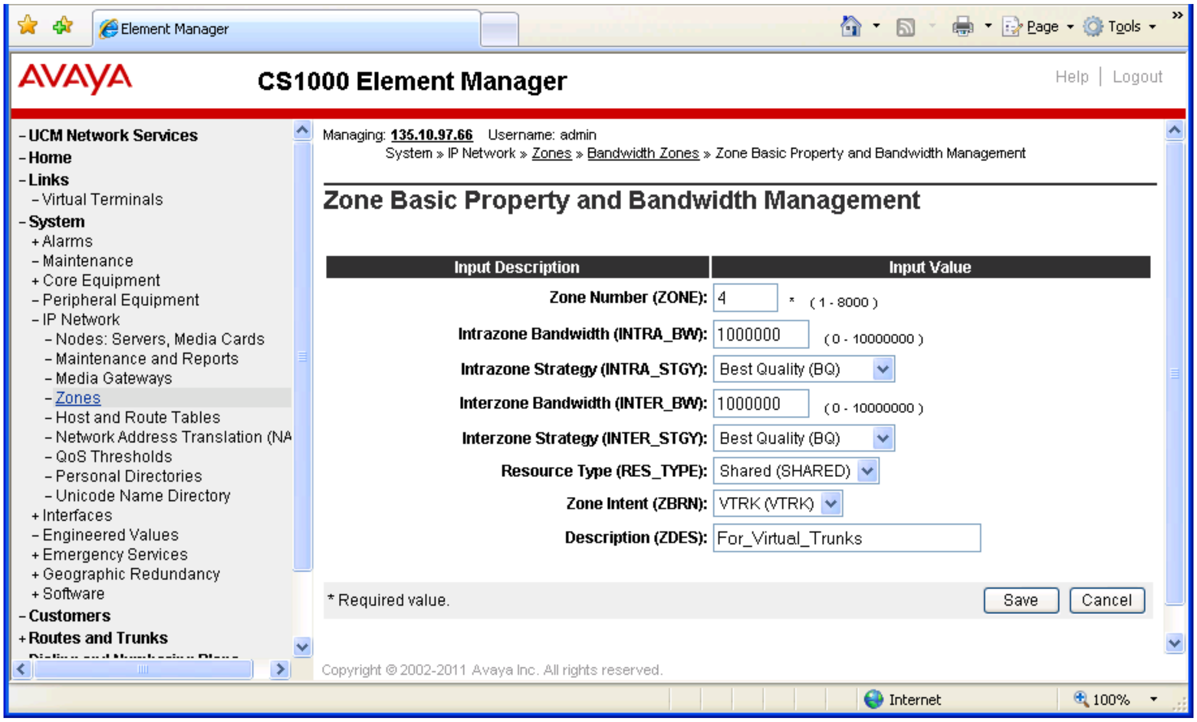

Figure 16: Zone Basic Property and Bandwidth Management

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.

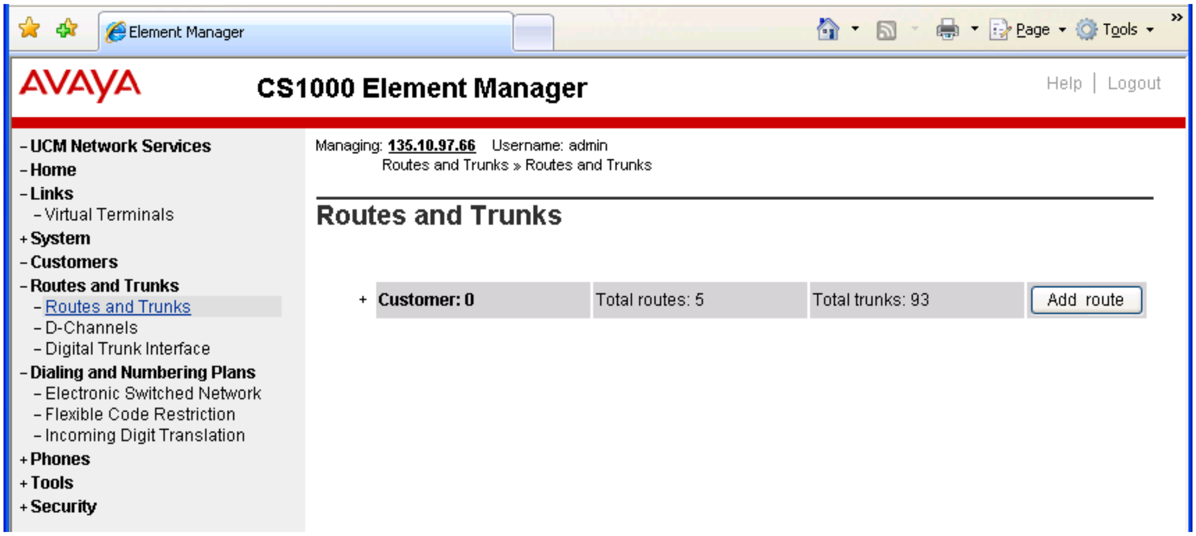

Figure 17: Routes and Trunks page

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**.

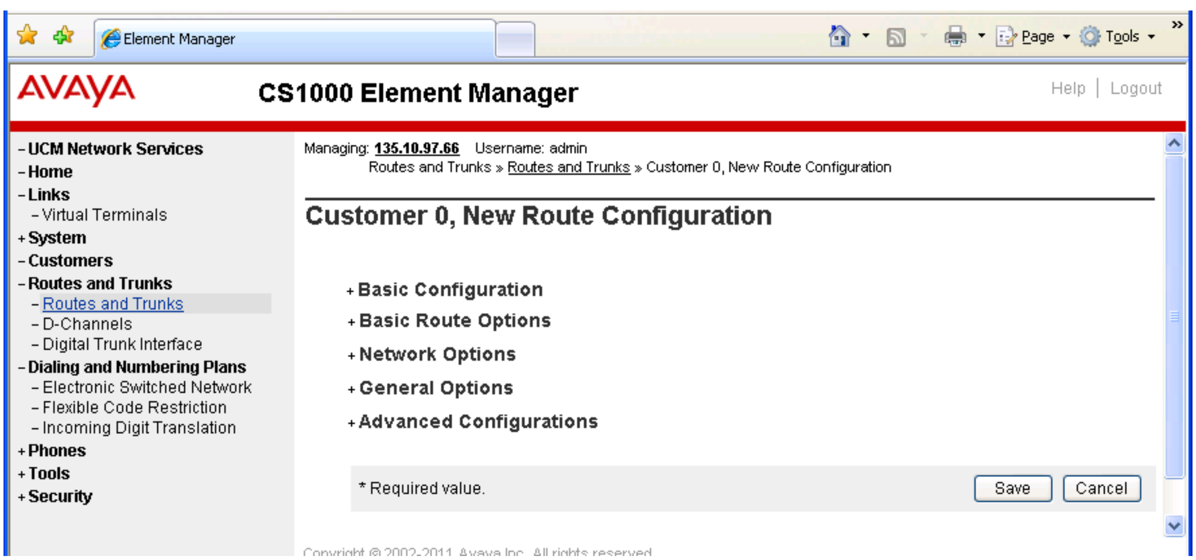

Figure 18: New Route Configuration page

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.

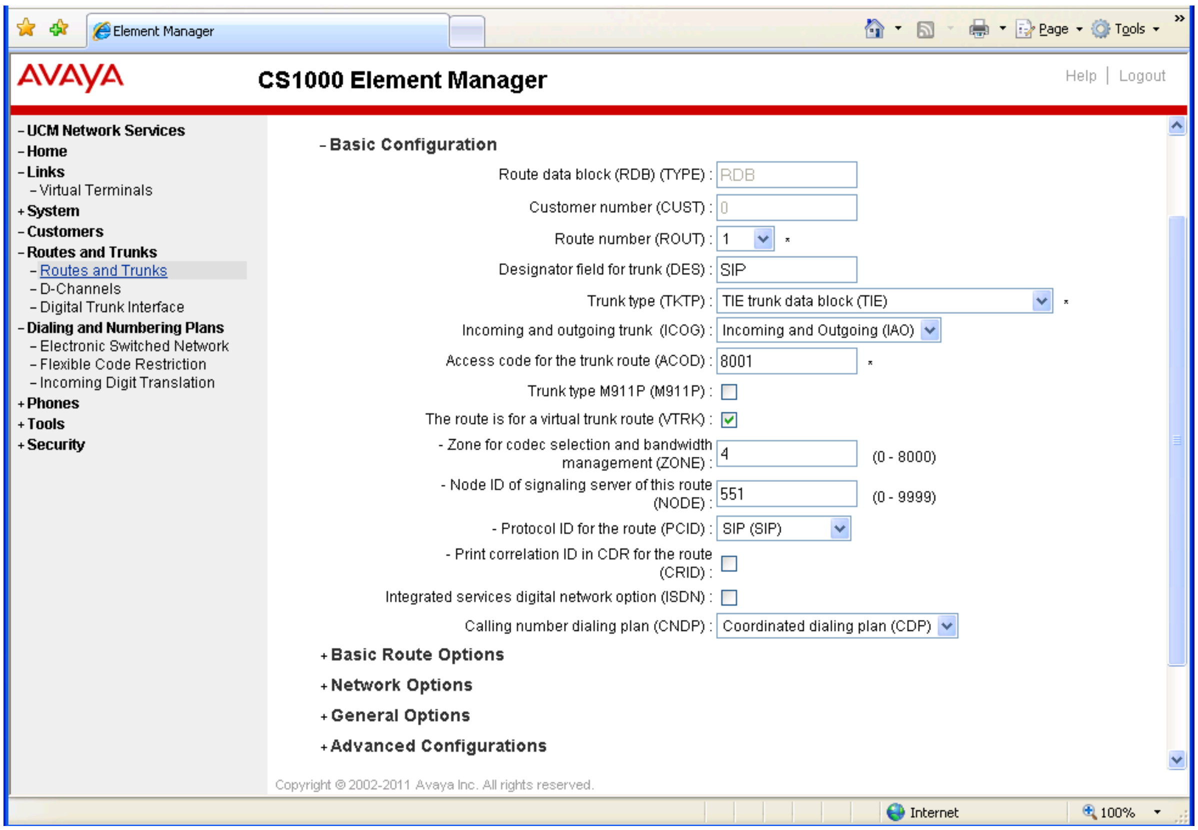

Figure 19: Basic Configuration section of new route

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.

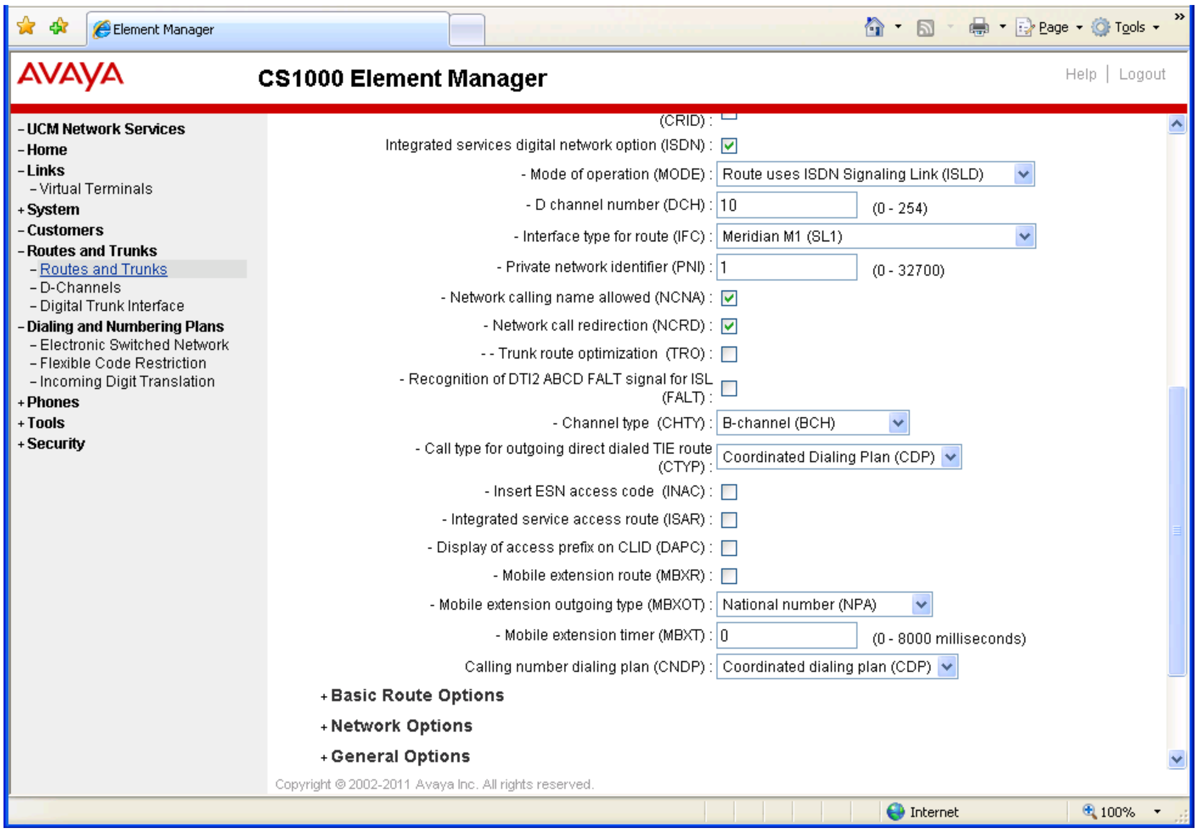

Figure 20: **Integrated services digital network option (ISDN) option page**

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

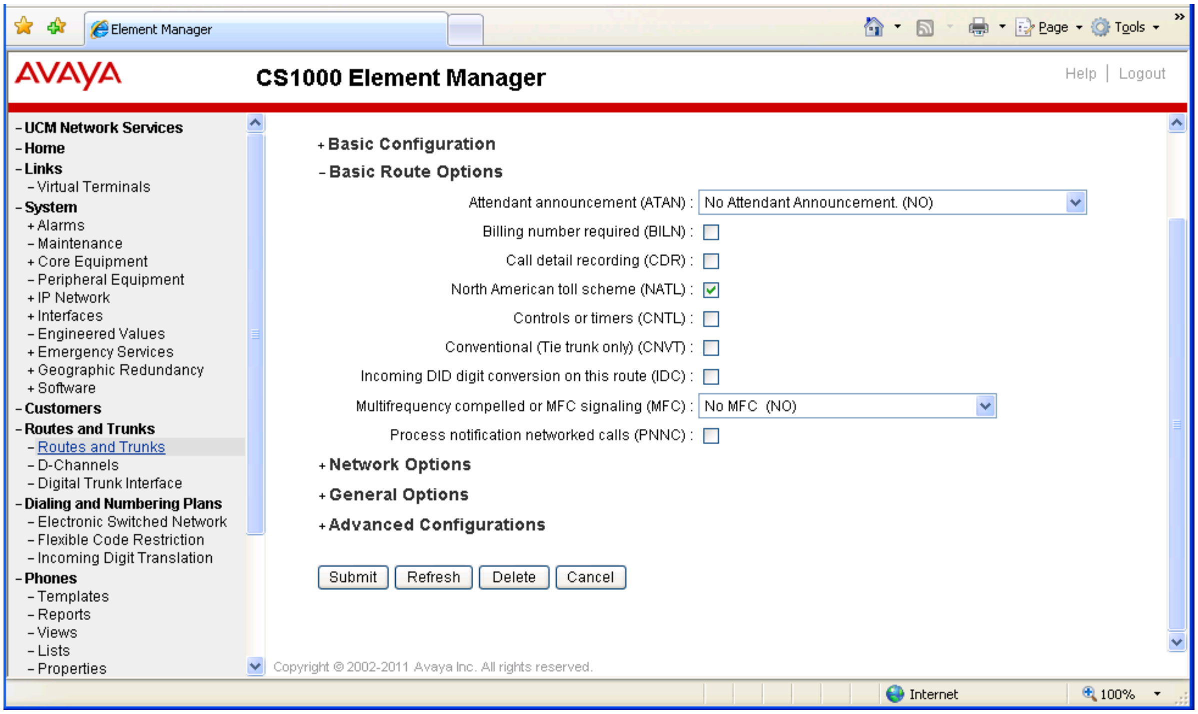

Figure 21: Basic Route Options of new Route

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

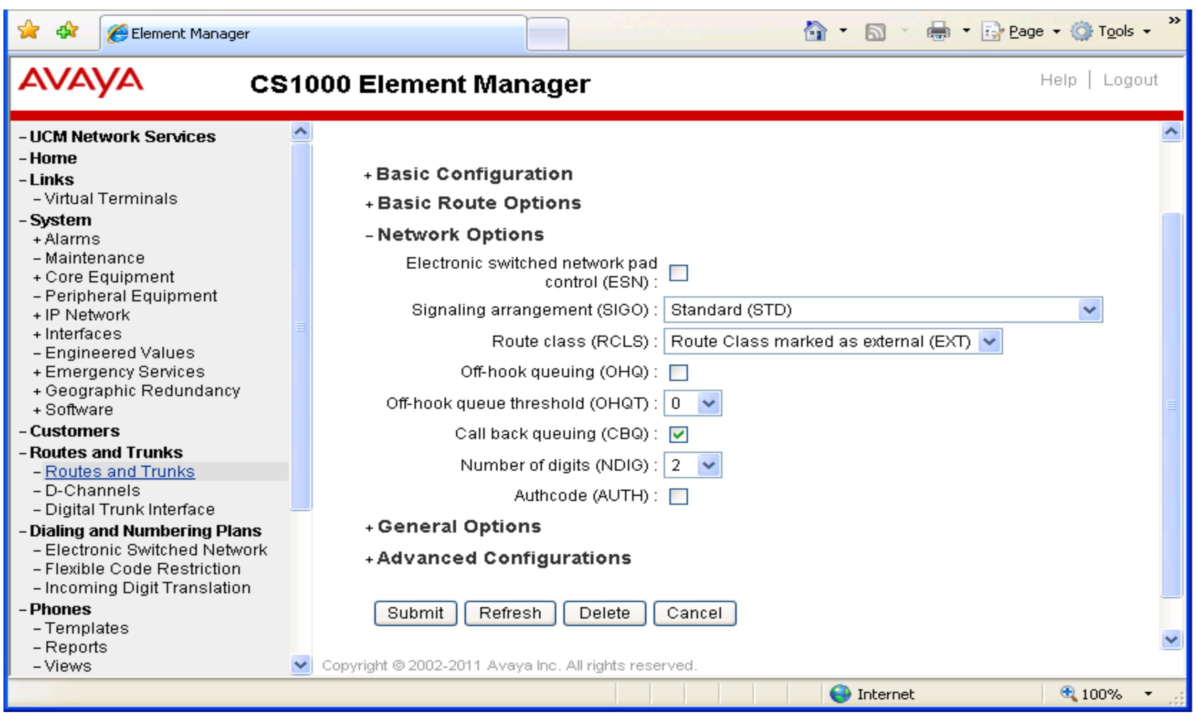

Figure 22: Network Options of new Route

Continue expanding the **General Options** section and keep default values as shown in Figure 23 below.

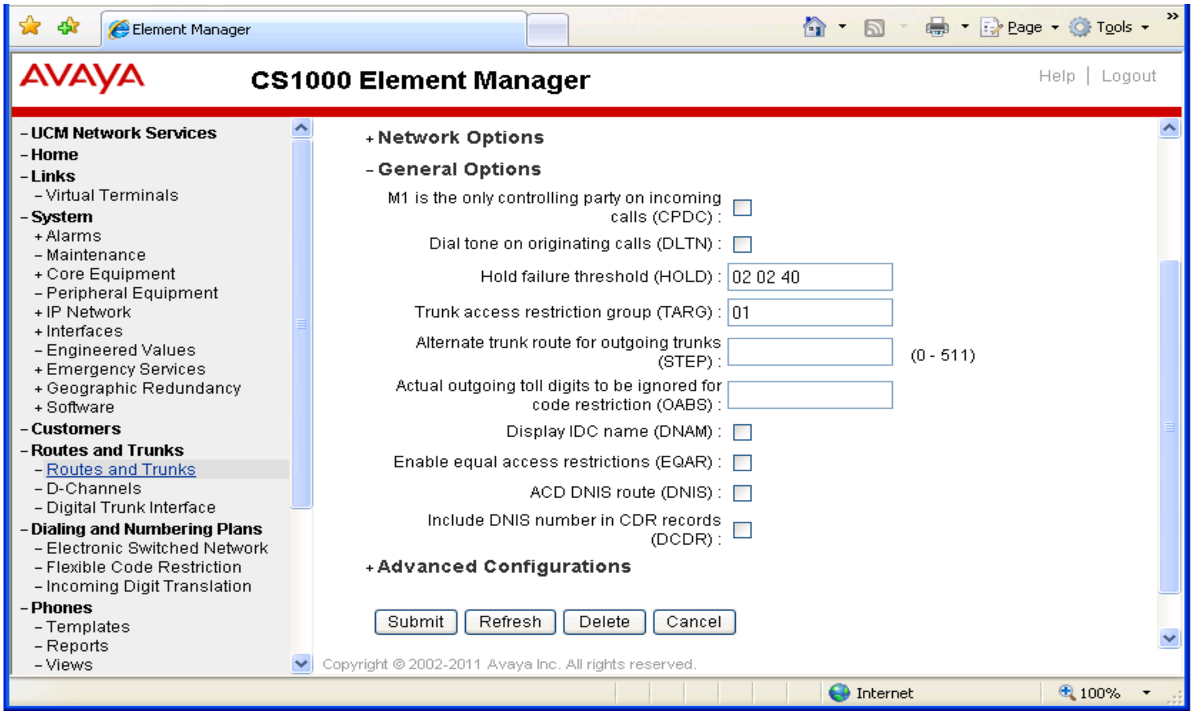

Figure 23: General Options of new Route

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

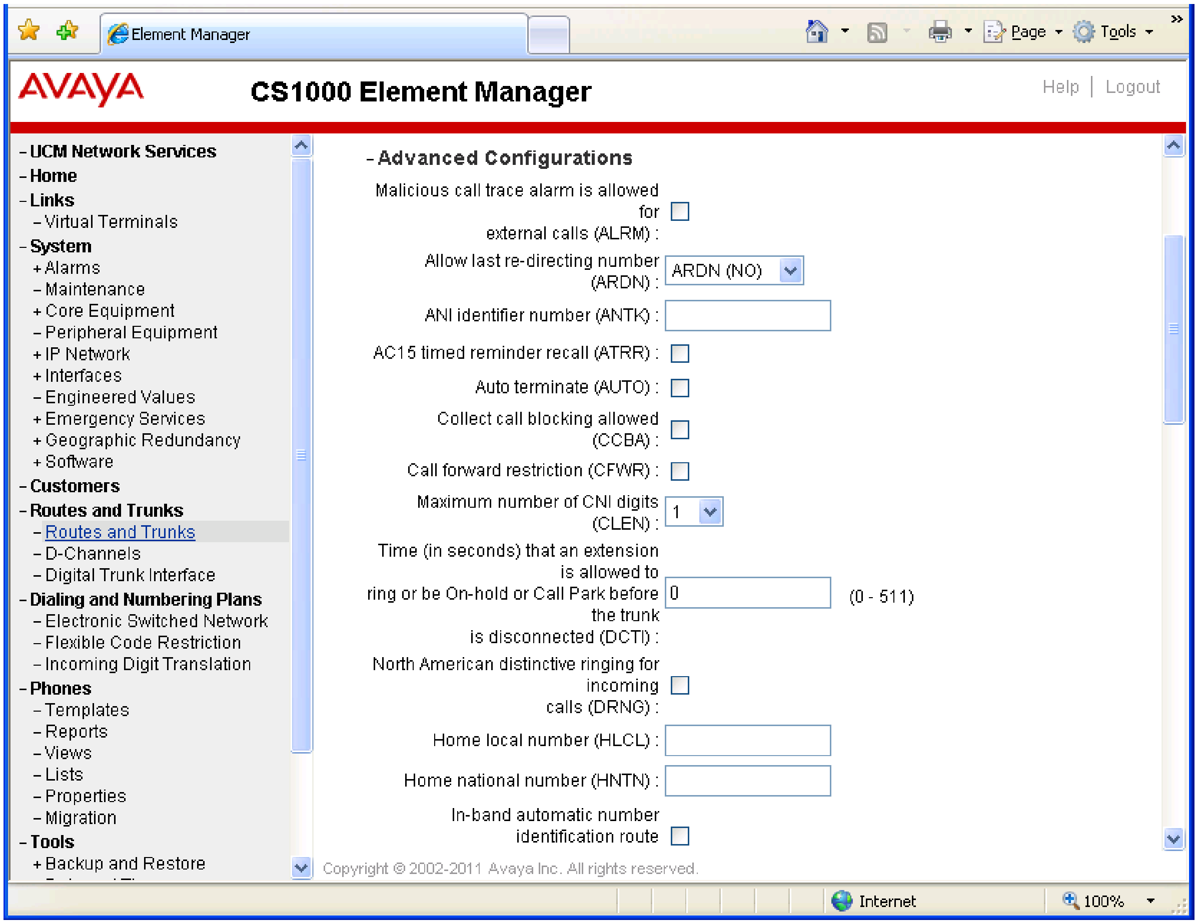

Figure 24: Advanced Configurations of Route

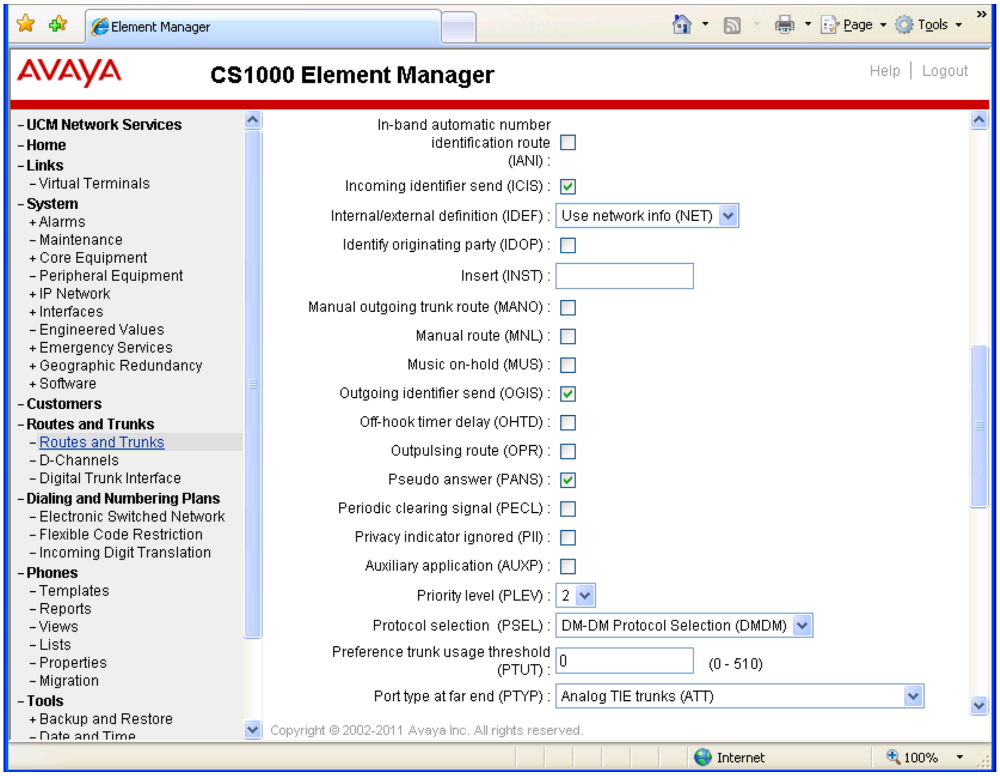

Figure 25: Advanced Configurations of new Route (cont)

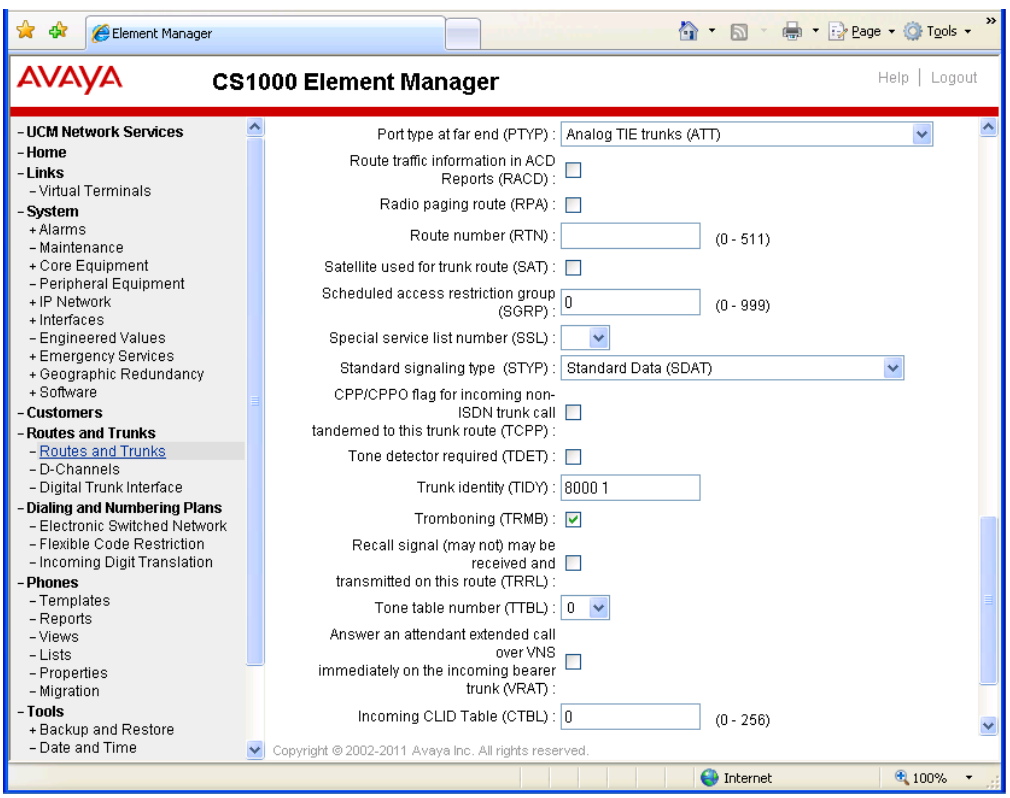

Figure 26: Advanced Configurations of new Route (cont)

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.

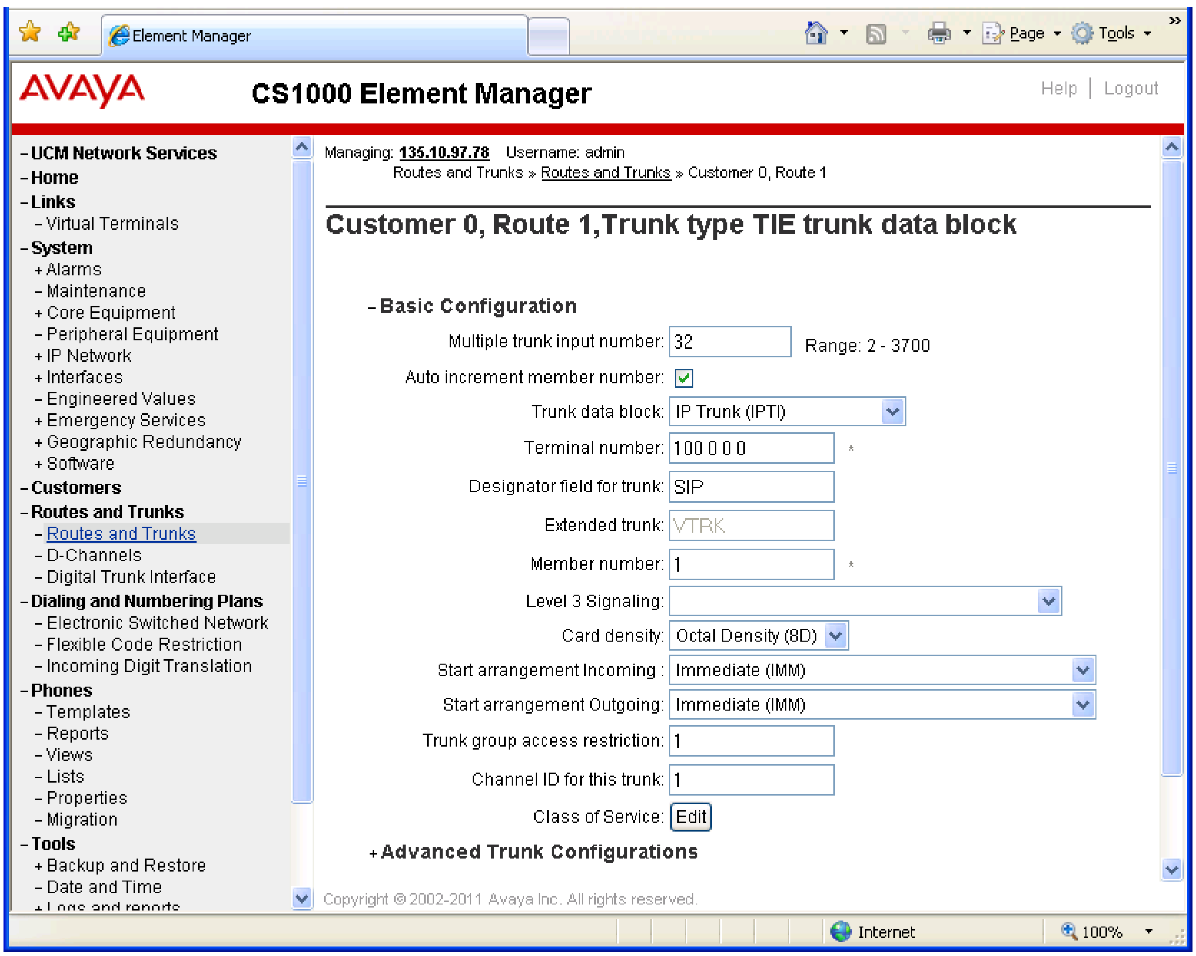

Figure 27: Basic Configuration of new Trunk

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.

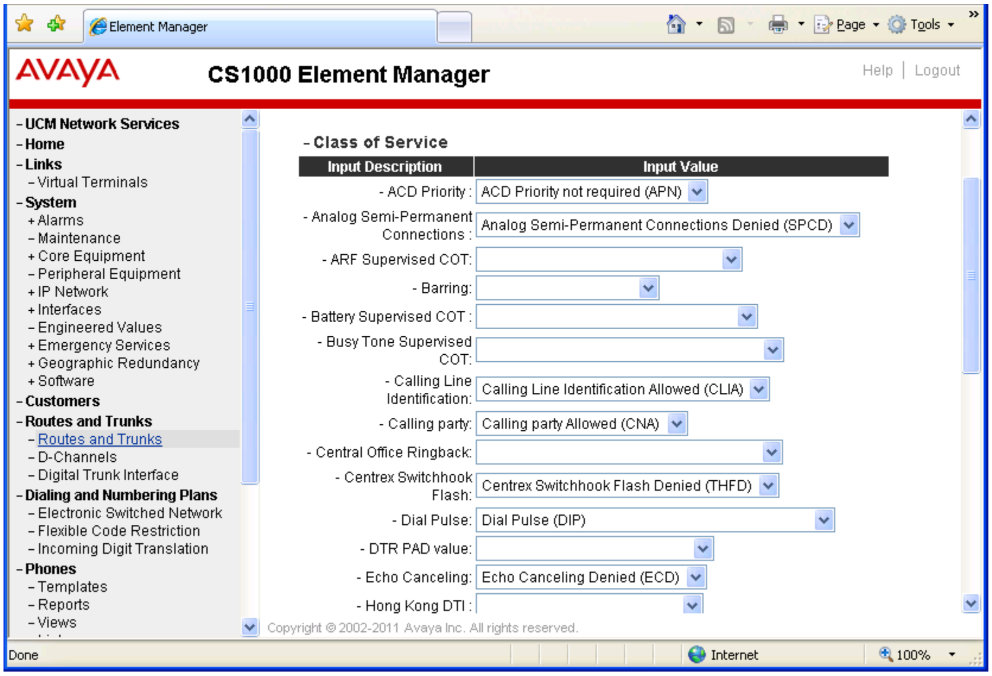

Figure 28: Class of Service of new Trunk

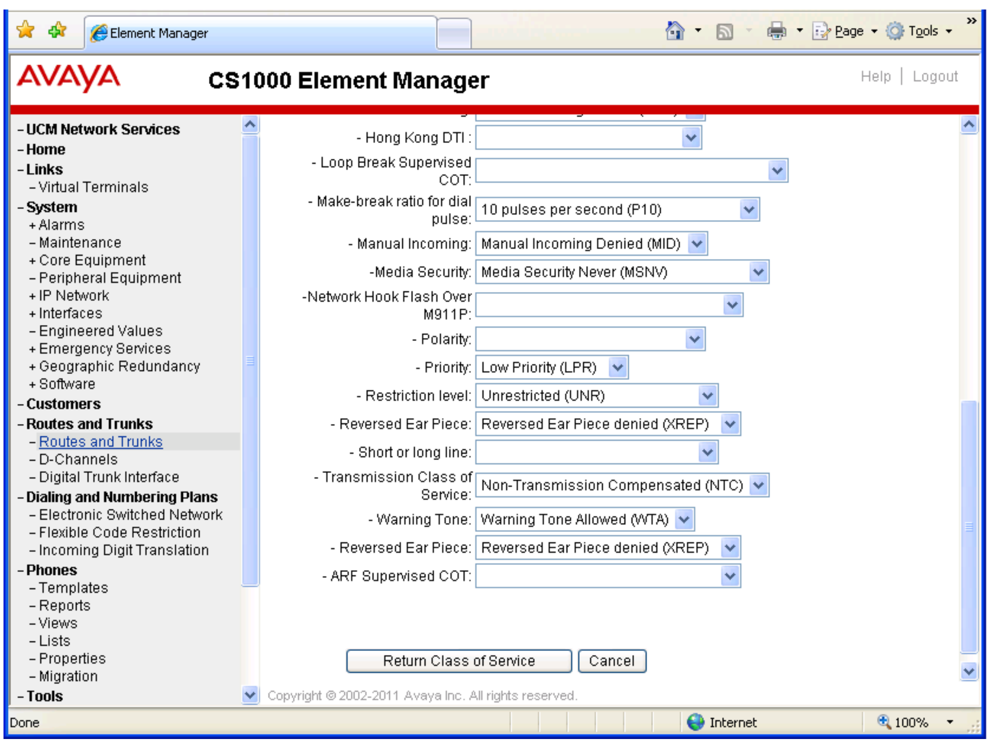

Figure 29: Class of Service of new Trunk (cont)

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.

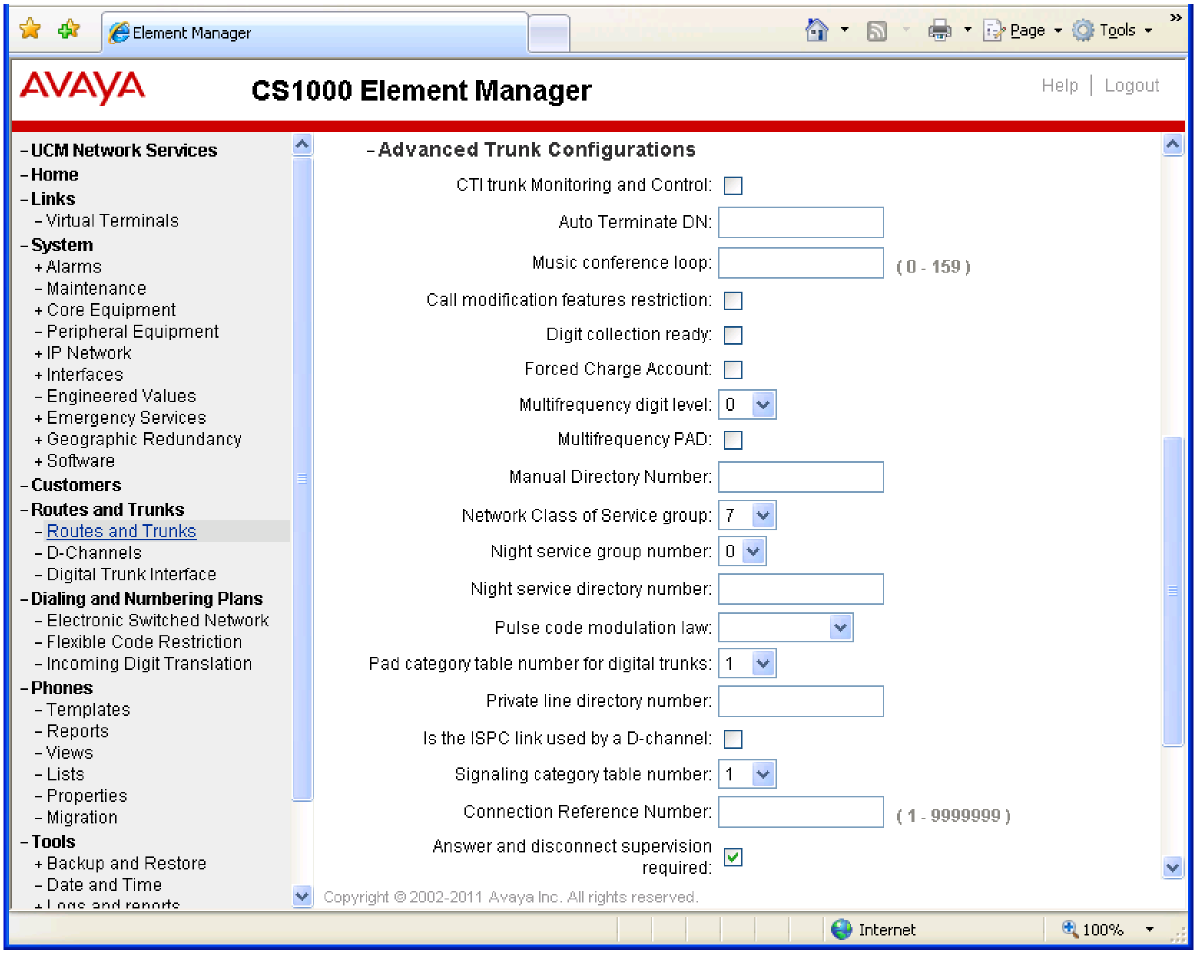

Figure 30: Advanced Trunk Configurations of new Trunk

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.

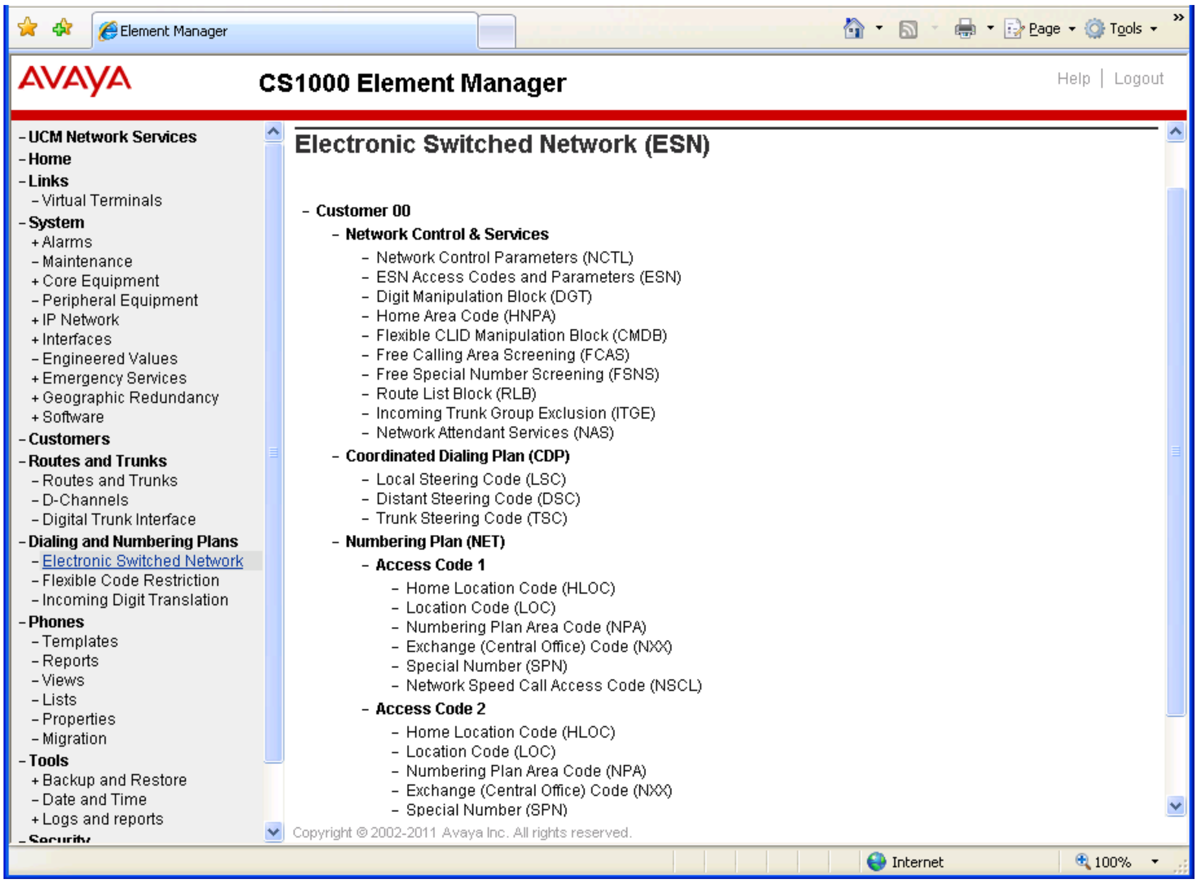

Figure 31: Electronic Switched Network (ESN) page

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.

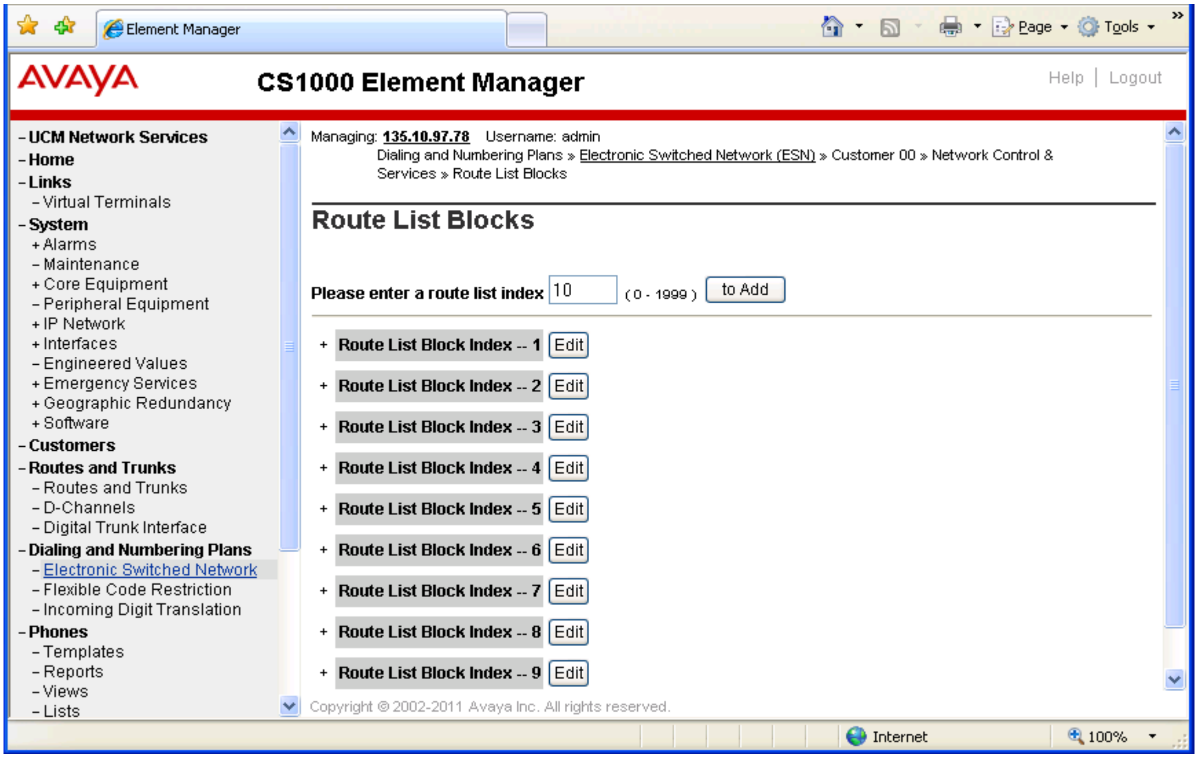

Figure 32: Route List Blocks page

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

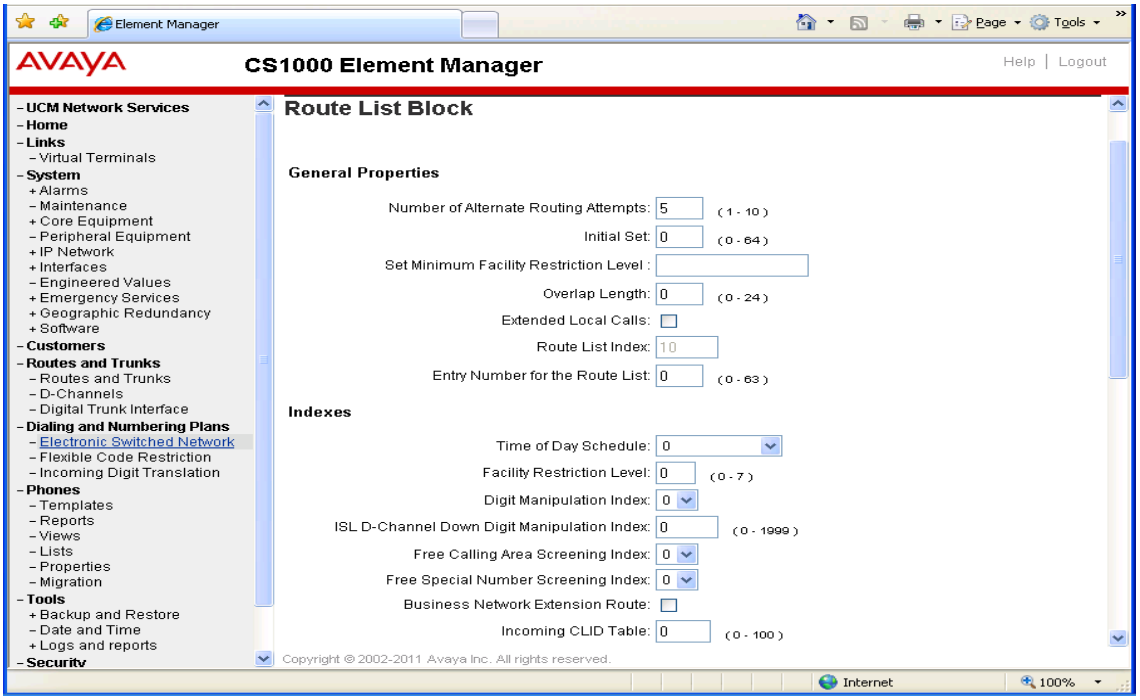

Figure 33: General Properties and Indexes of Route List Block page

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.

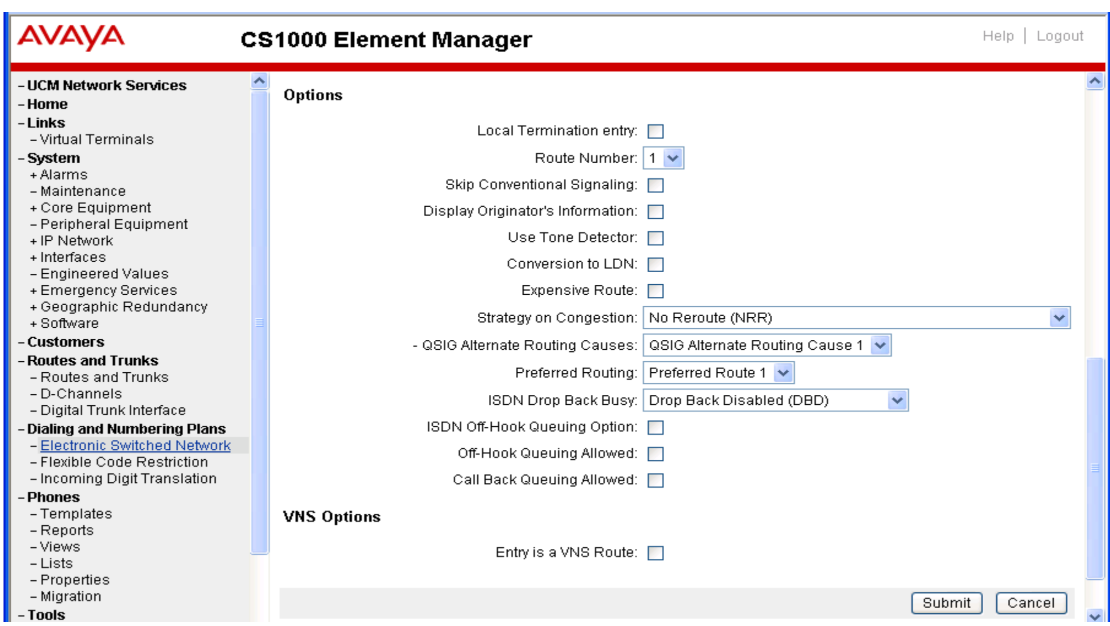

Figure 34: Options and VNS Options sections of Route List Blocks page (cont)

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.

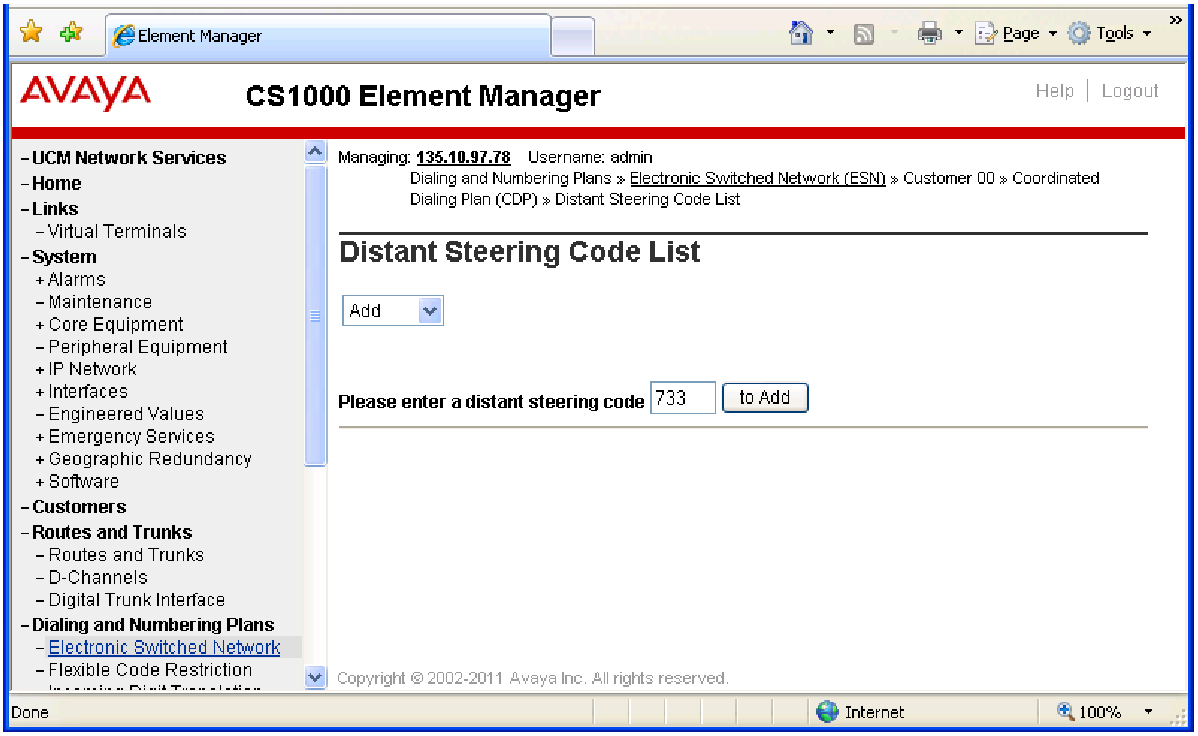

Figure 35: Distant Steering Code List page

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.

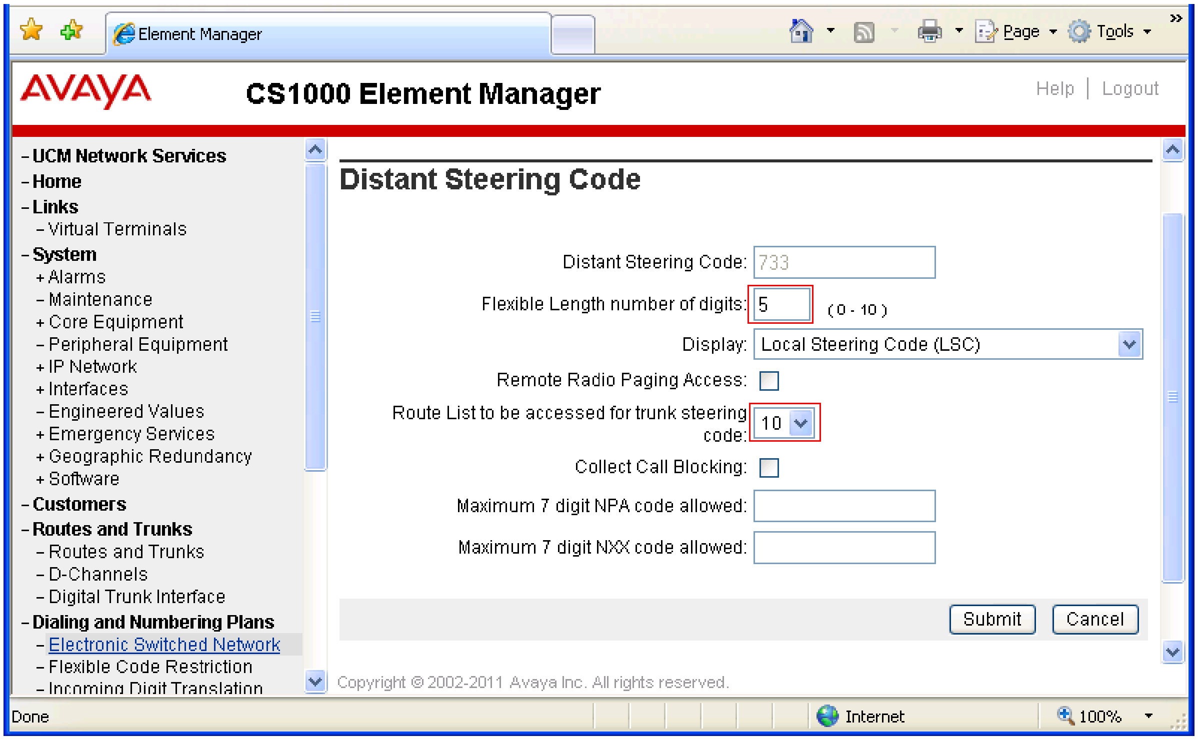

Figure 50: Distant Steering Code page

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 RCBD
    ICDD CDMD 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 EXR0
    USMD USRD ULAD CCBD RTDD RBDD RBHD PGND FLXD FTTC DNDY DNO3 MCBN
    FDSD 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 HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
     DRDD EXR0
     USMD USRD ULAD CCBD RTDD RBDD RBHD PGND FLXD FTTC DNDY DNO3 MCBN
     FDSD 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.

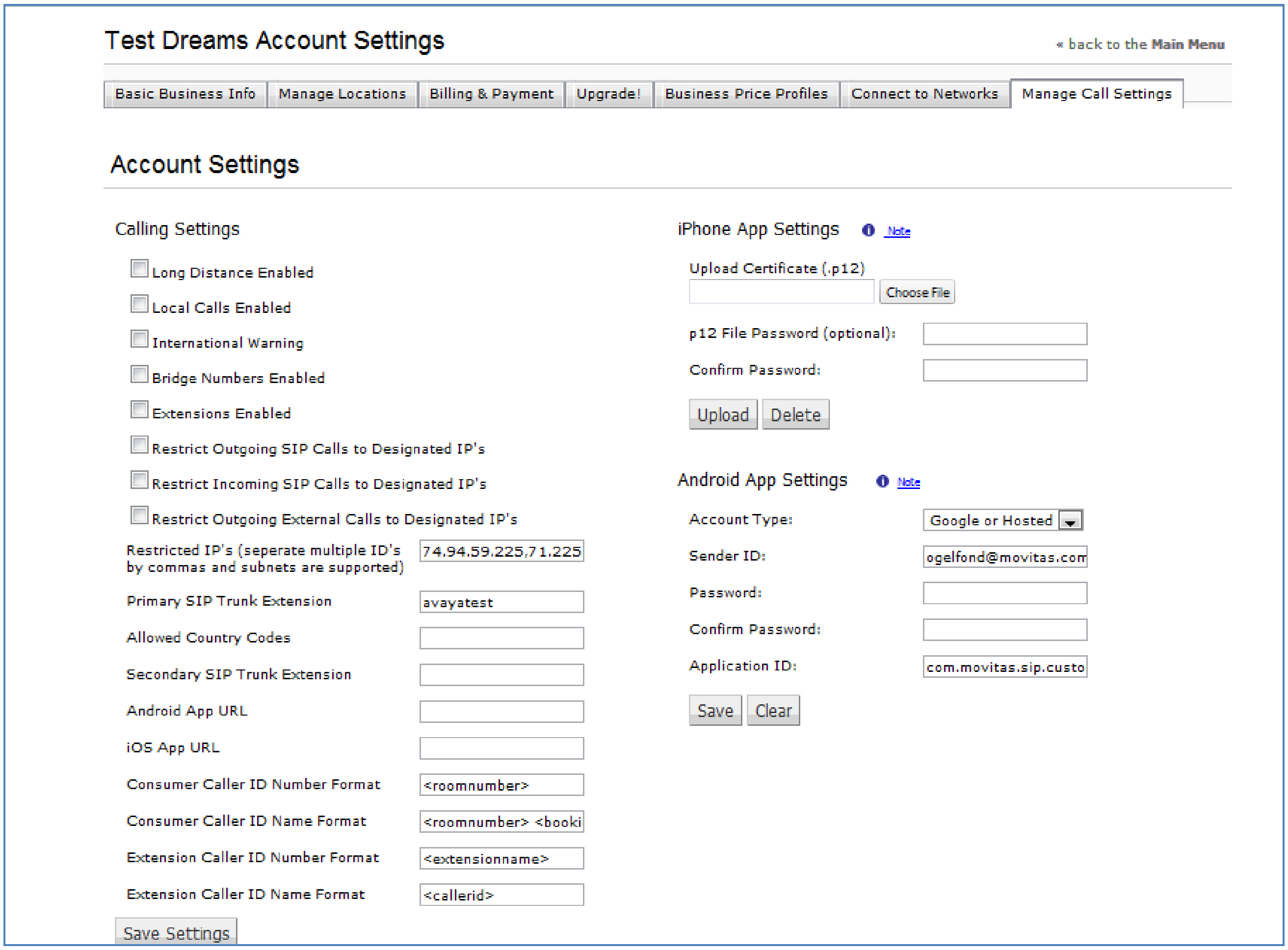

Figure 56: Dreams Account Settings page

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.

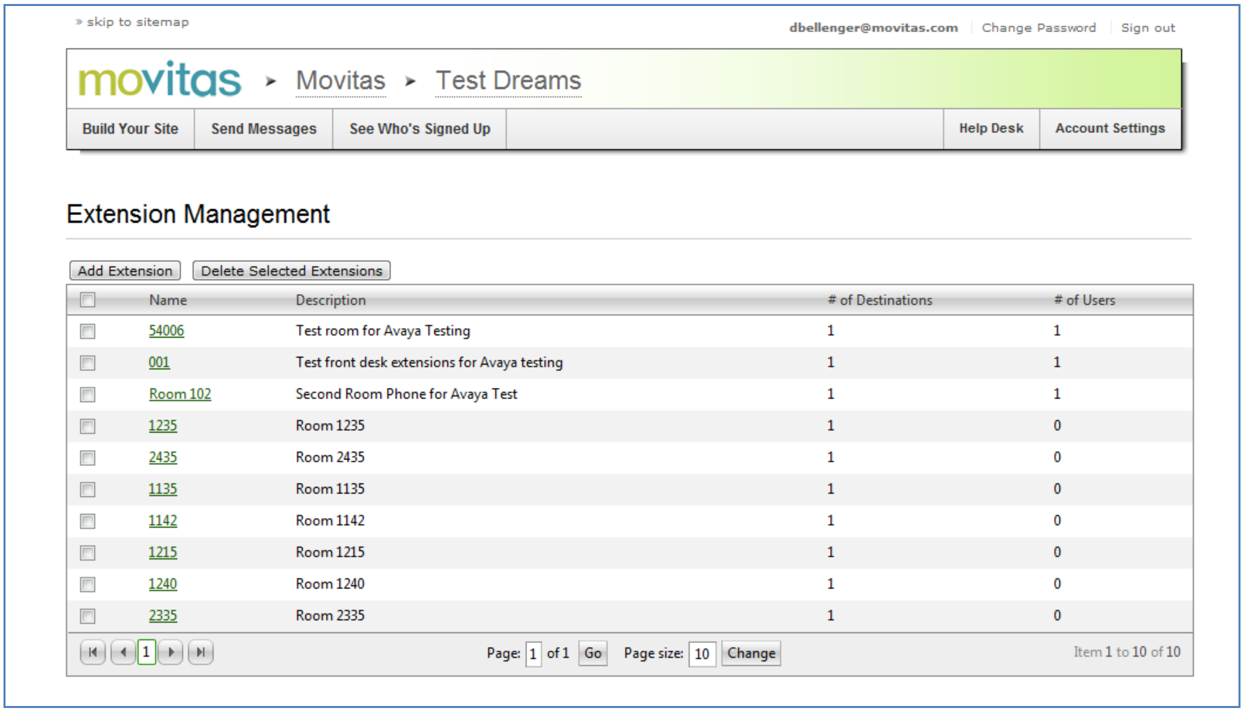

Figure 57: Setup Business Extensions

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

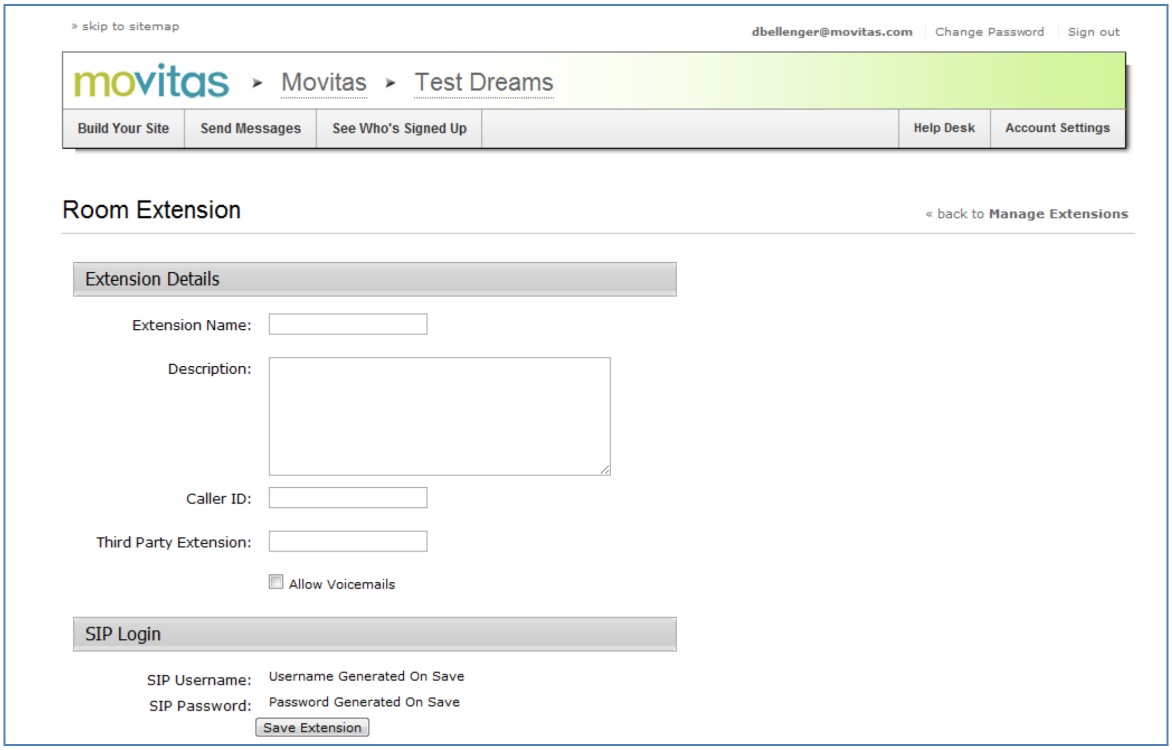

Figure 58: Setup Business Extensions (cont)

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.

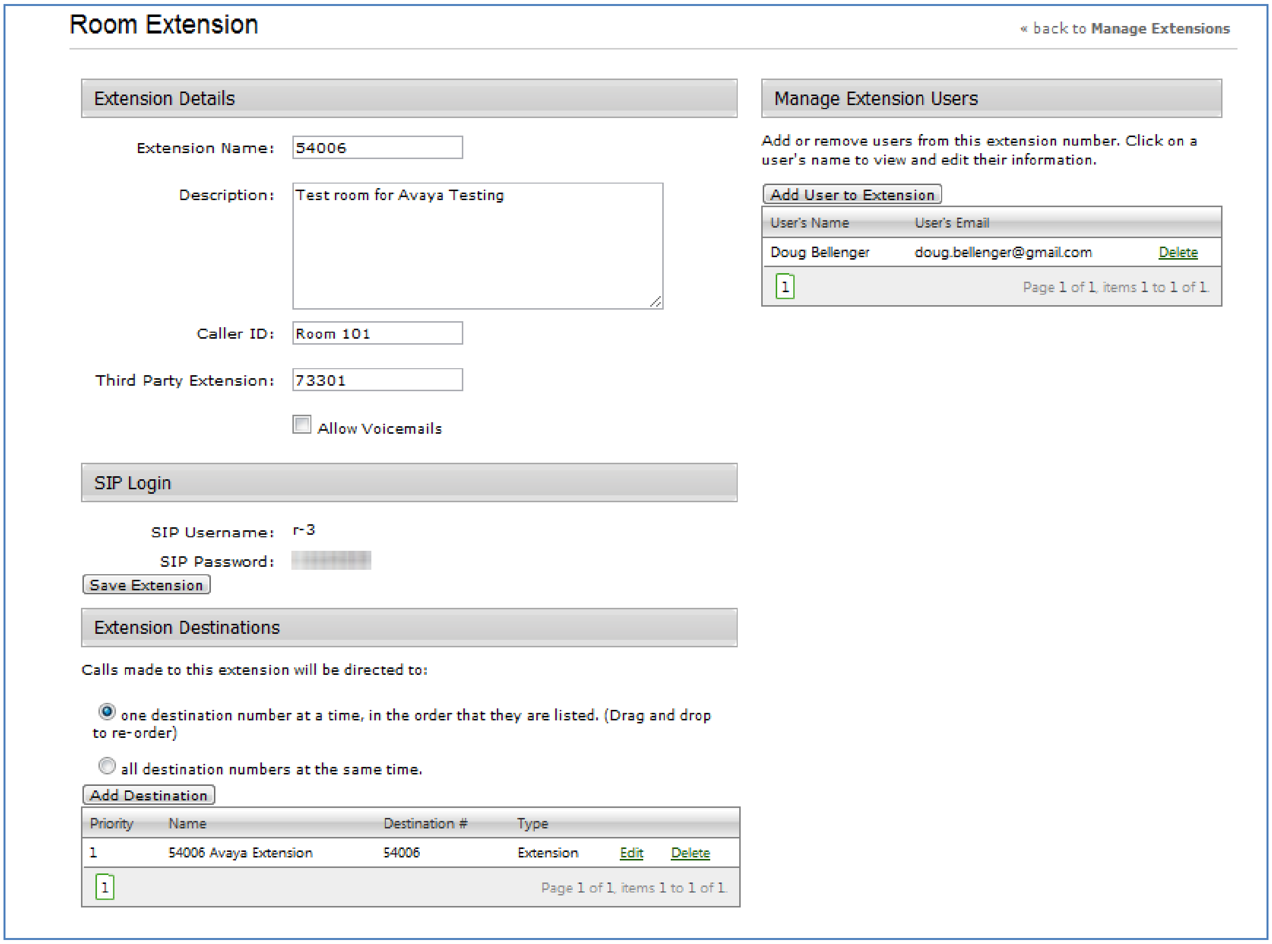

Figure 59: Setup Business Extensions (cont)

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.

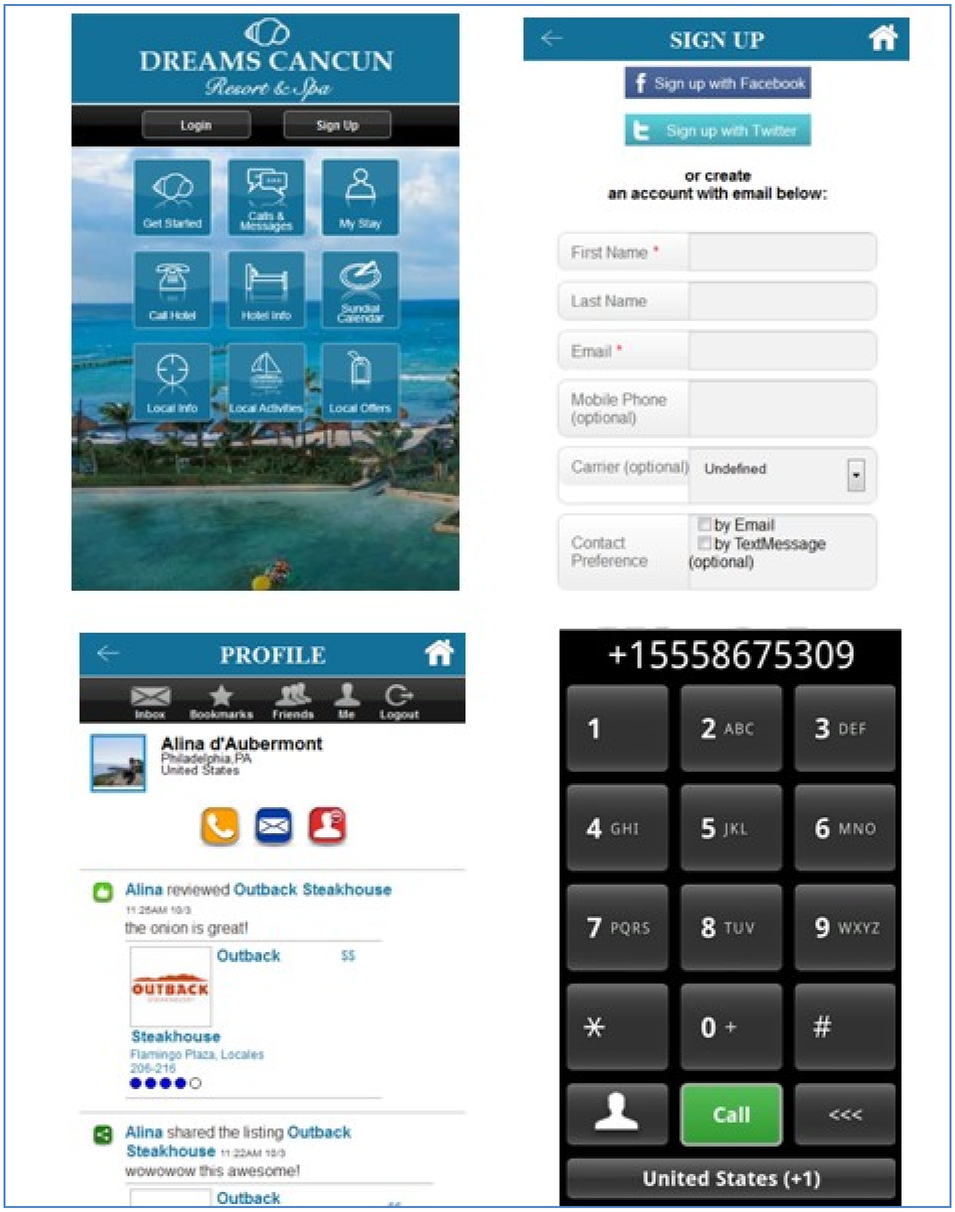

Figure 60: Sample of User create account on the application

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.

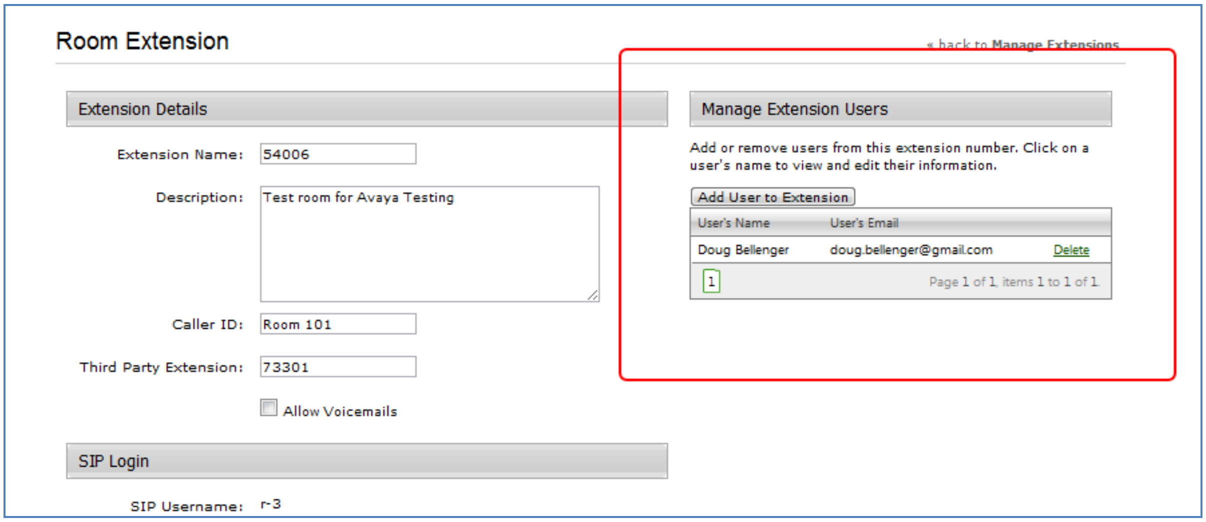

Figure 61: Assign Front Desk Extension to User

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.

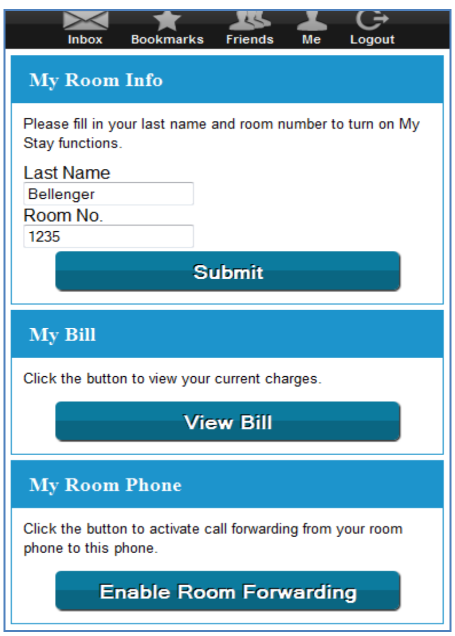

Figure 62: Assign Room Extension to User

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.

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