Avaya Solution & Interoperability Test Lab

Application notes for the Vocera Communication System Release 4.1SP6 with the Avaya Communication Server 1000 Release 6.0 – Issue 1.0

Abstract

These Application Notes describe the solution comprised of the Avaya Communication Server 1000 SIP Trunk Release 6.0 and the Vocera Communication System Release 4.1SP6. During the compliance testing, the Vocera Communication System Release 4.1SP6 was able to register to the SIP proxy Server of the Communication Server 1000 via SIP trunk. The Vocera Communication System was able to receive and re-direct the calls from the Communication Server 1000 IP and SIP Phones to Vocera B2000 Badges and vice versa. The compliance testing was focused on the interoperability of between the Avaya Communication Server 1000 and the Vocera Communication System.

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

These application notes provide detail configurations of the Avaya Communication Server 1000 release 6.0 (hereafter referred to as CS 1000) and the Vocera Communication System release 4.1SP6 (hereafter referred to as Vocera Server). During the compliance testing, the Vocera Server was tested to make sure all telephony features properly functioned and interoperated with CS 1000 via SIP trunk.

1.1. Interoperability Compliance Testing

The focus of this testing was to verify that the Vocera Server was able to interoperate with the CS 1000 system. The following areas were tested:

- SIP Trunk is established successfully between the Vocera Server and the CS1000.
- Basic calls between the Vocera Server and different telephone types of CS1000 (SIP, non-SIP and emulated PSTN telephones).
- DTMF transmission.
- Conference and Transfer calls from different telephone types of the Avaya CS1000 (SIP, non-SIP and emulated PSTN telephones) to the Vocera Server clients (wireless badge B2000) and vice versa.
- Call Forward (All Call, No Answer and Busy) and Call Forward to voicemail with Message Waiting Indication (MWI) notification.
- Other telephony features: Busy, Hold and Retrieve calls.

1.2. Support

For technical support on the Vocera Server, please contact Vocera technical support at the website http://www.vocera.com/about/support.aspx or telephone: +1 408-882-5100 or email support@vocera.com for more detail.

2. Reference Configuration

Figure 1 illustrates the test bed configuration used during the compliance testing between the Avaya CS 1000 and the Vocera Server.

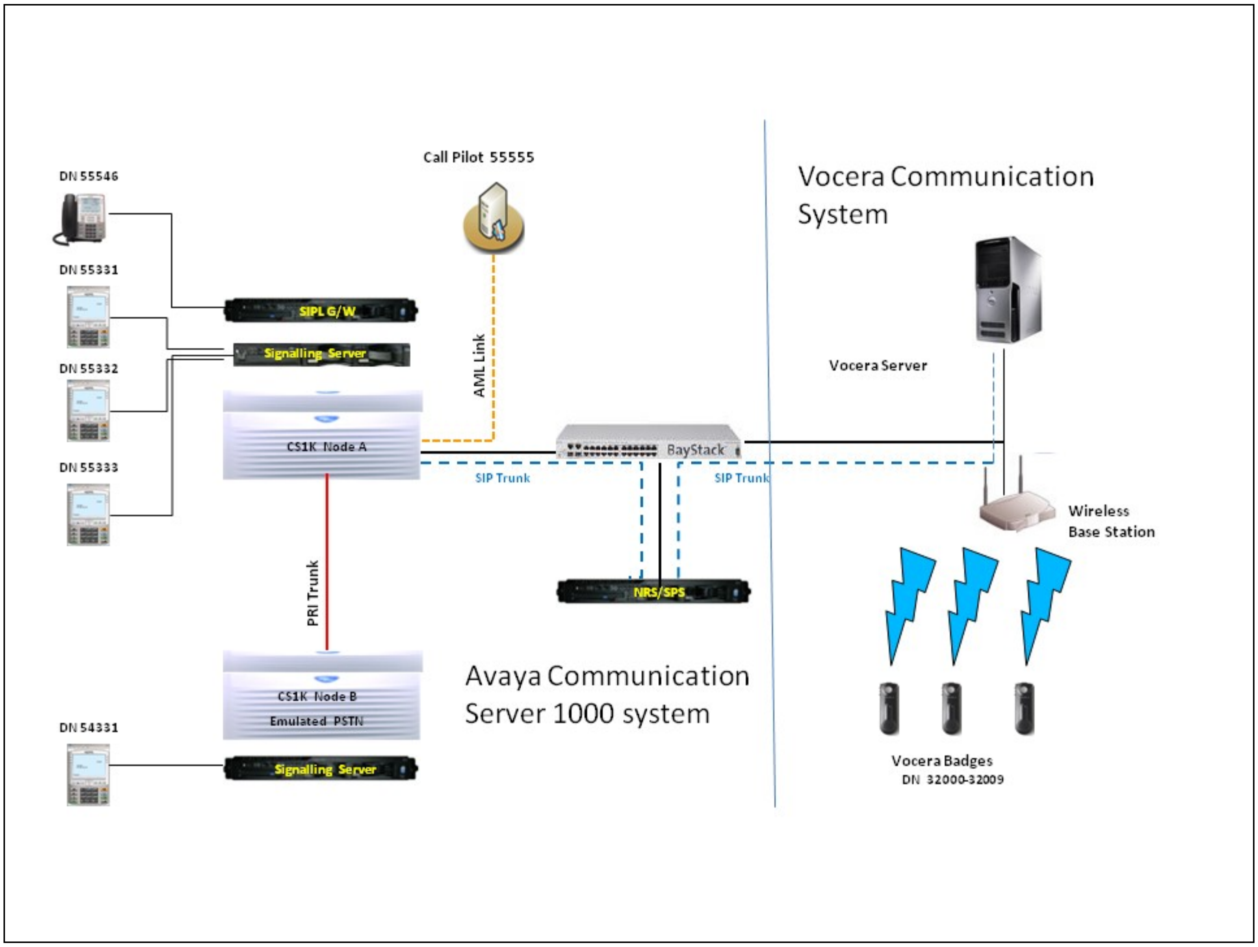

Figure 1: Network Configuration

3. Equipment and Software Validated

System	Software Version
Avaya CS 1000	• Call Server (CPPM): 6.00 • Signaling Server (CPPM): 6.00.18 • SIP Line Gateway (HP DL320): 6.00.18
Avaya CallPilot voicemail system	• 5.0
Vocera Communication System	• Vocera Server 4.1SP6 • Wireless Badge B2000 • Wireless Base Station

4. Configuring Avaya CS 1000

This section describes the details on how to configure the Avaya CS 1000 SIP Trunk using the Element Manager. The command line interface (CLI) option is also available to provision the SIP Trunk application on the CS 1000 system if user chooses to do so.

Prerequisites:

- A CS 1000 server which has been:
  - Installed with CS 1000 Release 6.0 Linux Base.
  - Joined CS 1000 Release 6.0 Security Domain.
  - Deployed with SIP Trunk Application.

  For more information on CS 1000 installation, maintenance, and upgrades, see section 9.

- The following software packages are enabled in the keycode.
- If any of these features have not been enabled, please contact your Avaya account team or Avaya technical support at http://www.avaya.com.

Package Mnemonic	Package Number	Package Description	Package Type (New or Existing or Dependency)	Applicable Market
SIP	406	SIP Service package	New package	Global
FFC	139	Flexible Feature Codes	Existing package	Global
SIPL_NORTEL	415	Nortel SIP Line package	Existing package	--
SIPL_3RDPARTY	416	Third-Party SIP Line Package	Existing package	--

4.1. Logging on to the Unified Communications Management (UCM) and Element Manager (EM)

This section provides the steps of how to log on the UCM Common Services of the CS 1000.

Using the Microsoft Internet Explorer 6.0260 or later to access the UCM by addressing the IP address or FQDN (Full Qualified Domain Name) of the UCM and then input the username/password which was defined during the primary security server setup.

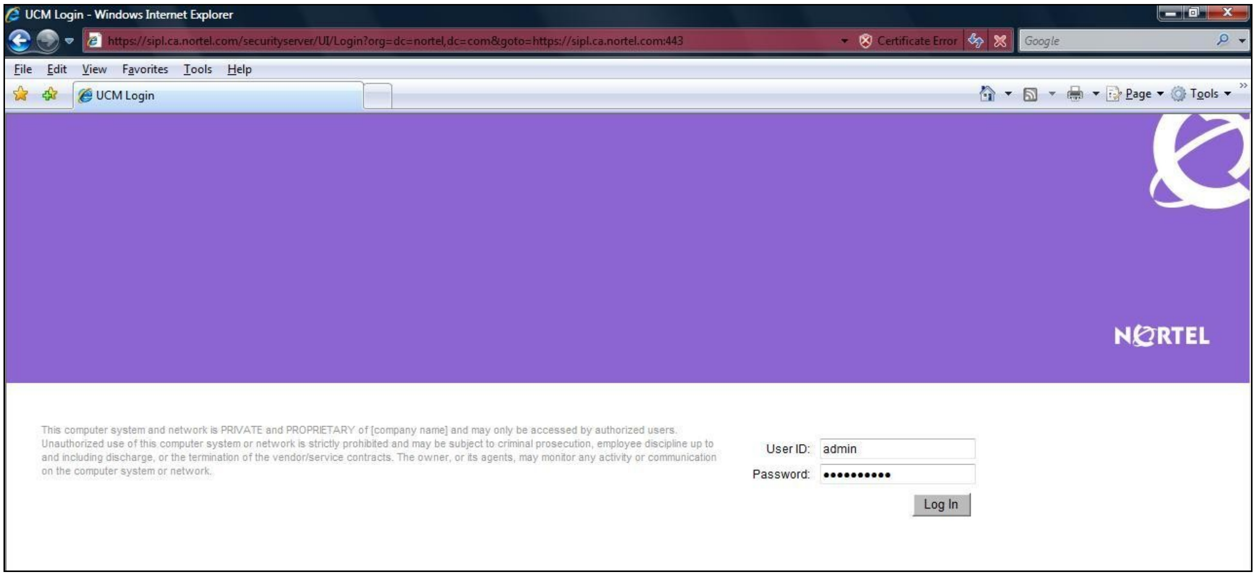

Figure 2: UCM Login web page

4.2. Configuring CS 1000 Network Routing Service (NRS) SIP Proxy Server

IP Peer Networking enables customers to distribute the functionality of CS 1000 systems over a Wide Area Network, using either Avaya SIP or H.323 Gateways, or third-party SIP or H.323 Gateways.

The Network Routing Services (NRS) Manager, a web-based management application, is used to configure, provision, and maintain the NRS. The NRS supports both SIP Proxy and SIP Redirect but only SIP Proxy was used for the compliance testing. Therefore, only SIP Proxy is mentioned in this document.

4.2.1. Creating a new SIP gateway endpoint for the Vocera Server on the CS 1000 NRS Manager

From the UCM homepage, navigate to the list of servers under Element names attribute; select the server **NRSM on SPS1** as shown in Figure 3.

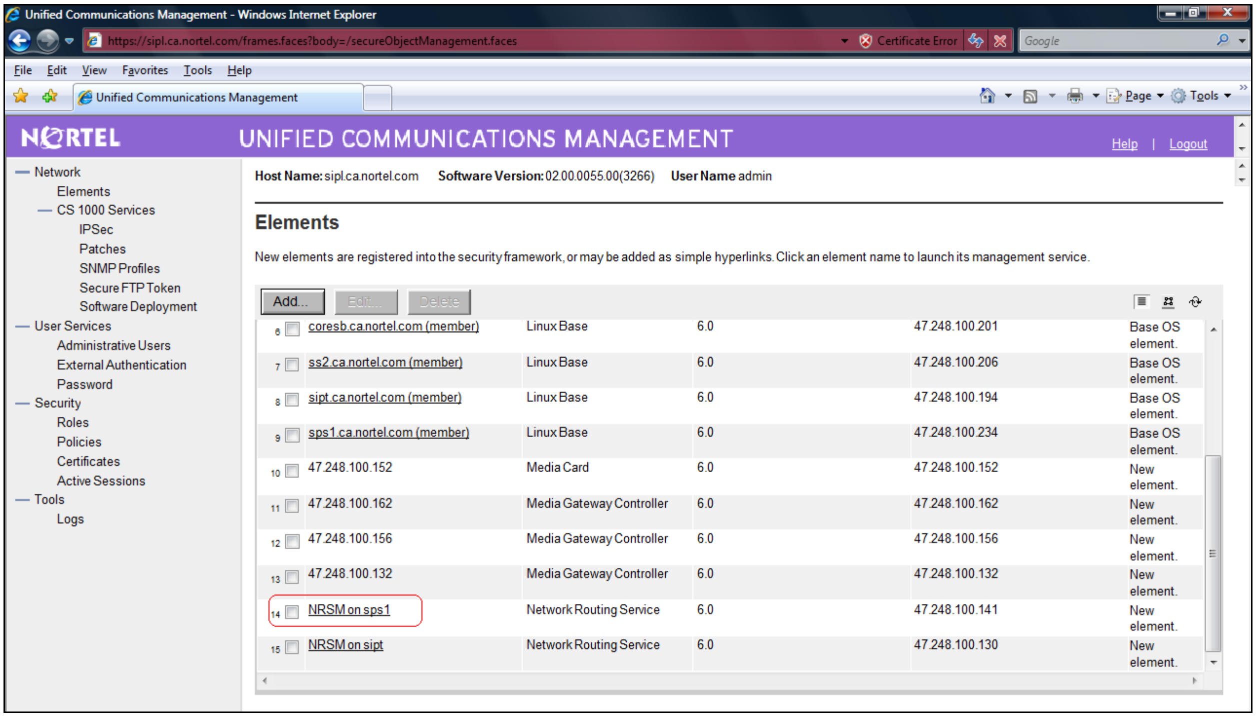

Figure 3: UCM Navigator Homepage

The NRS Manager page appears as shown in Figure 4.
To add a gateway endpoint, follow the steps below:

- Select the **Standby database** mode of operation by clicking on the radio button.
- On the left hand column menu, choose **Number Plans** -> **Endpoints**
- From the attribute **All Service Domains** pull down list, choose domain name **nortel-dplab.com**.
- From the **All L1 domains,** choose **UDP** from the pull down menu.
- Similarly, for the **All L0 domains** choose **CDP**.
- Then click on the **Add** button.

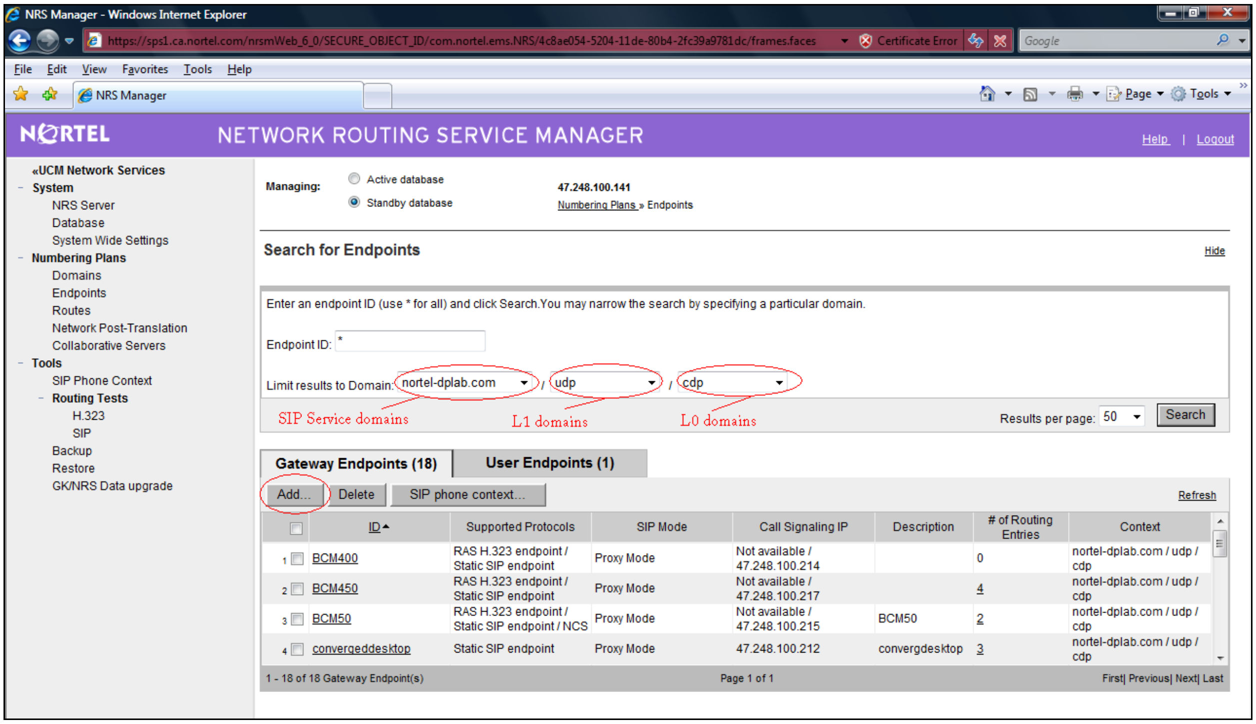

Figure 4: Adding a Gateway Endpoint

The configuration details of the gateway endpoint page will appear as shown in Figure 5.

To configure a gateway endpoint, all the highlighted attributes in red circles should be altered as shown in Figure 5. The Static Endpoint Address attribute should be filled in with the Vocera Server IP address (not shown). Others are left at default values.

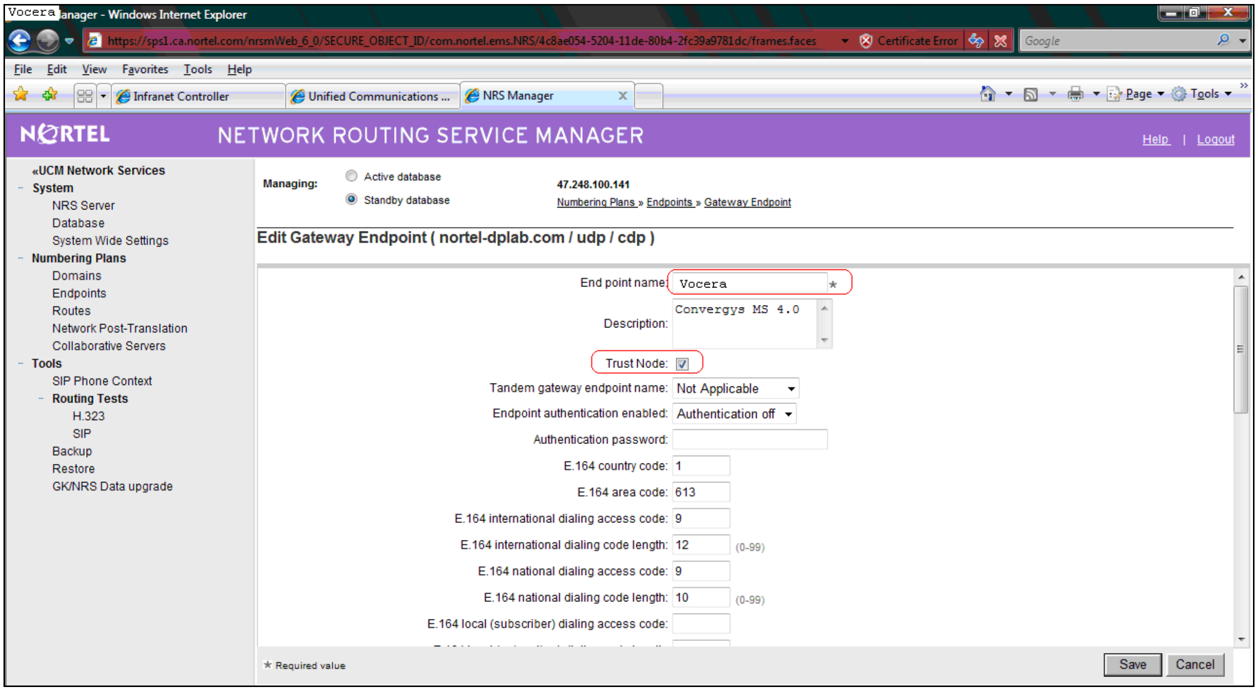

Figure 5: Input Fields of the Vocera Gateway Endpoint

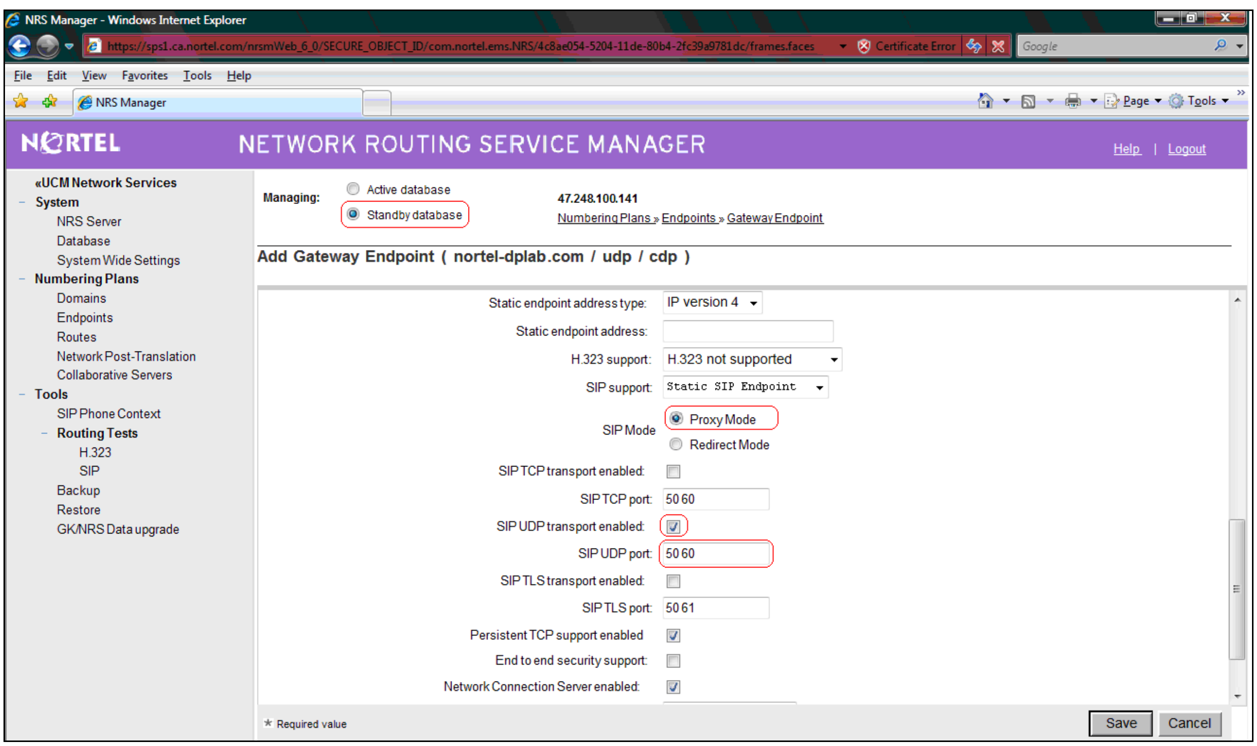

Figure 5b: Input Fields of the Vocera Gateway Endpoint (Continued)

Click on the **Save** button in order to complete the newly created gateway endpoint for Vocera. Select the **Database** on the left column under the **System** menu, the **Database** page will appear as shown in Figure 6. Click on the **Cut over** button to transfer configured data of the gateway endpoint from the **Standby database** to the **Active database**.

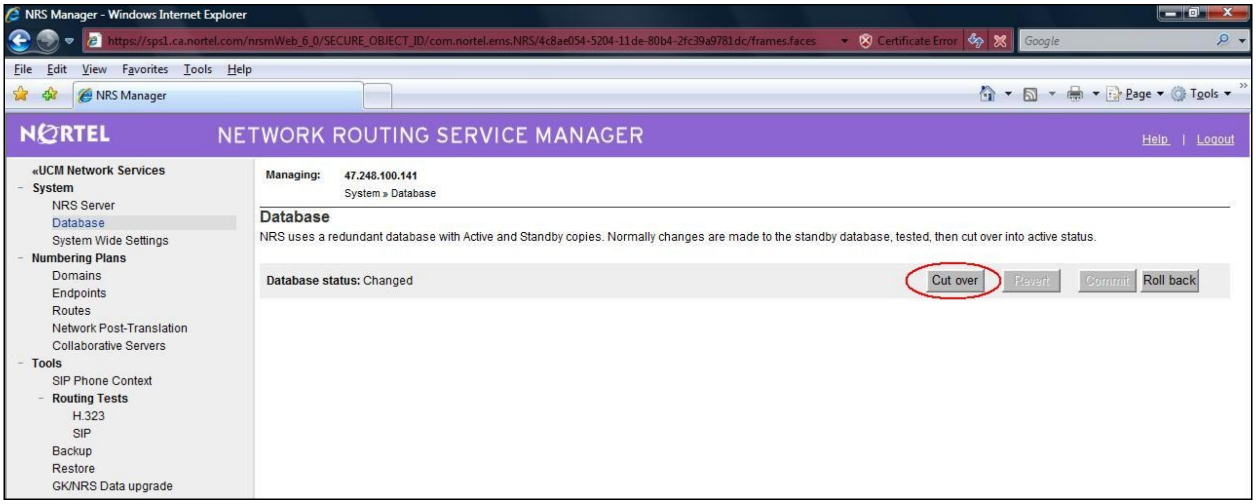

Figure 6: Cut over for Database on NRS

Then click on the **Commit** button to active the changes on the NRS.

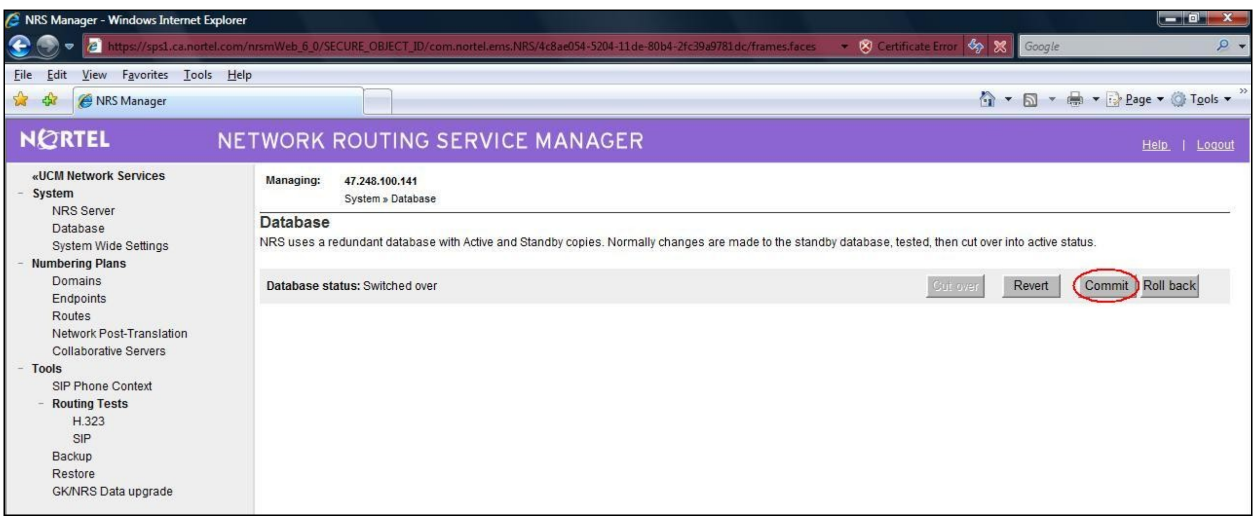

Figure 7: Commit the Database of NRS

4.2.2. Creating a routing entry for Vocera gateway endpoint on the NRS

To create a route entry for the **Vocera** gateway endpoint, follow the steps below:

- Select the **Standby database** mode of operation by clicking on the radio button.
- On the left hand column menu, choose **Number Plans** -> **Routes**
- From the attribute **All Service Domains** pull down list, choose domain name **nortel-dplab.com**.

- From the **All L1 domains,** choose **UDP** from the pull down menu.
- From the **All L0 domains,** choose **CDP** from the pull down menu.
- From the **Endpoint Name** pull down list, select the **Vocera** endpoint name in the list of gateway endpoints.
- Then click on the **Add** button.

Note: The **Vocera** endpoint name only shows up on the list of gateway endpoints when its L1 and L0 domains are selected.

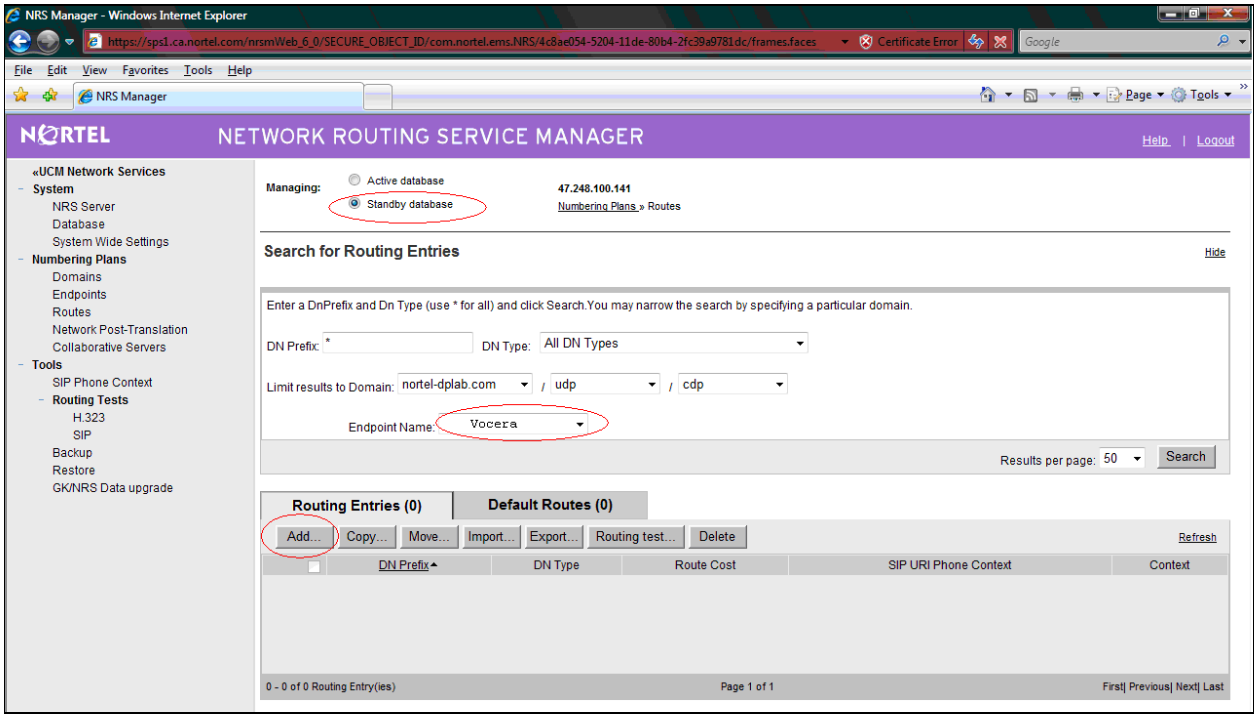

Figure 8: Adding Route Entry for Vocera Endpoint

To configure a routing entry, all the highlighted attributes in red circle should be altered as shown in Figure 9, then click on **Save** to complete.

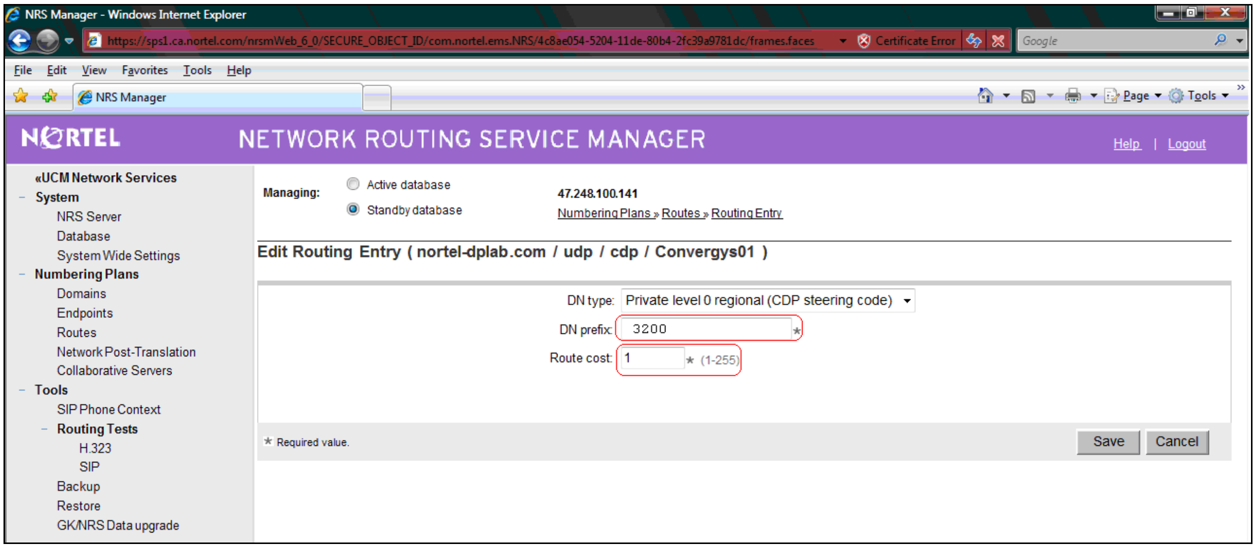

Figure 9: Configure Route Entry for Vocera Endpoint

Similarly in creating the Vocera gateway endpoint, the process of the **Cut over** and **Commit** on the database of the NRS needs to be applied to active the new route. Figure 10 below shows the new route **3200** of the **Vocera** endpoint.

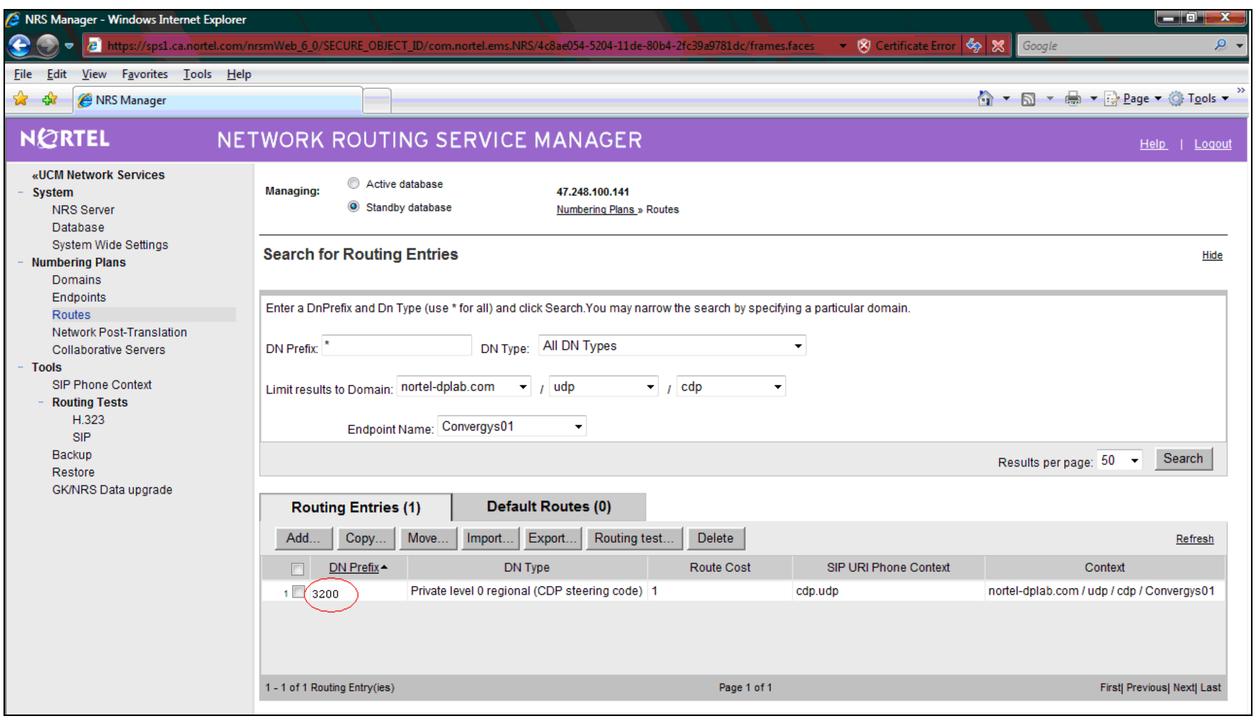

Figure 10: Route Entered for Vocera Endpoint

4.3. Configuring CS 1000 Call Server by Element Manager (EM)

This section describes the steps on how to create:

- Route Data Blocks
- Trunks
- ESN Data Block
- Patching

To allow calls to be route to and from CS 1000 Call Server to Vocera Server.

4.3.1. Registering CS Node ID to the NRS

Perform the following steps to register CS 1000 Node ID **555** to the NRS

To launch the EM of Call Server, log on the UCM and then click on the Element Name **EM on SS2** link of the Call server that needs to be configured as shown in Figure 11 below.

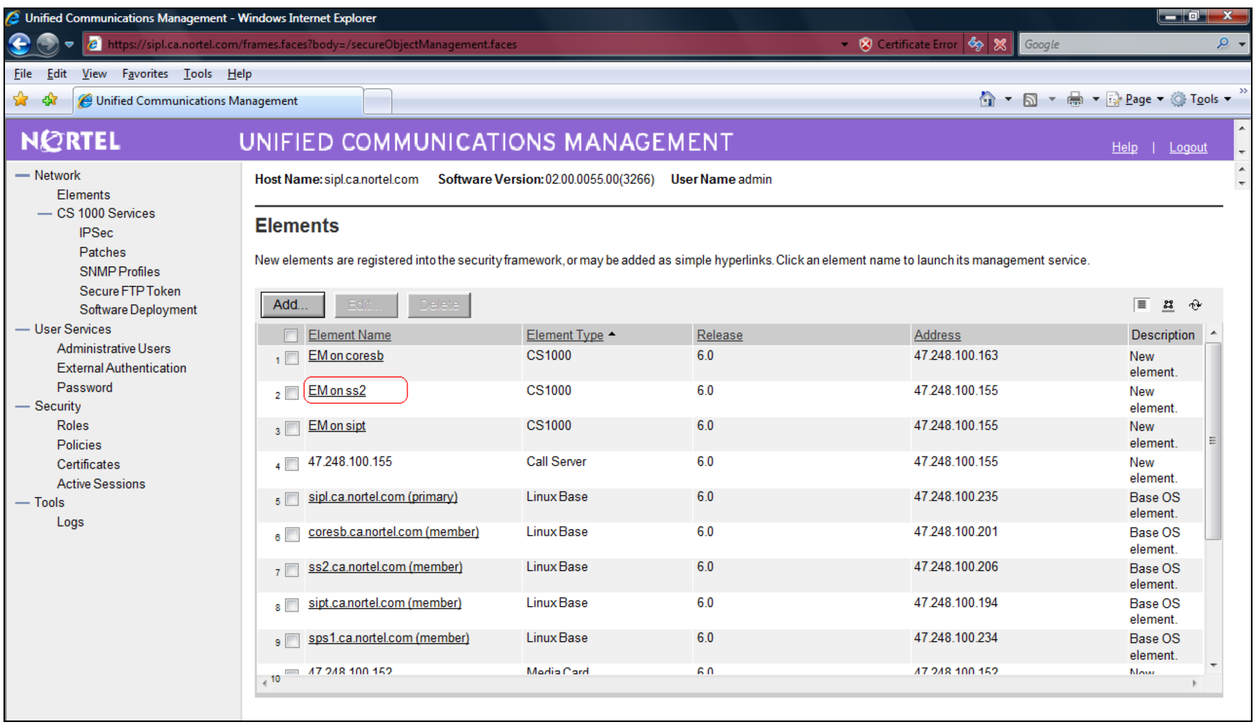

Figure 11: EM of Call Server on the UCM.

The EM homepage of Call Server appears as shown in Figure 12.

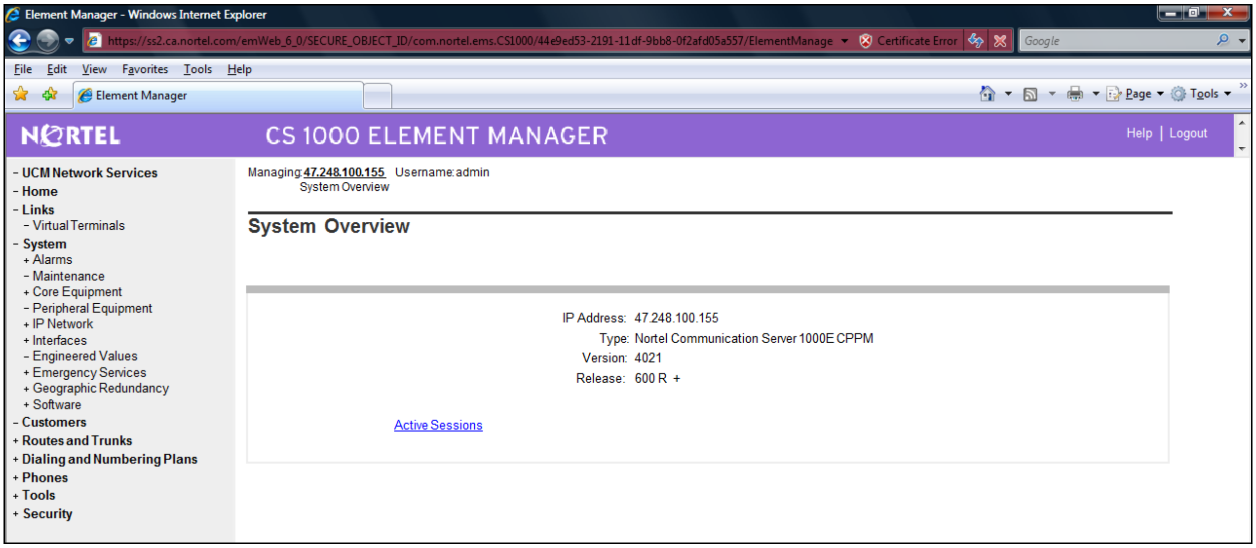

Figure 12: Element Manger homepage of Call Server

To register CS Node ID to the NRS follows steps below:

- From the left menu column of the EM page, navigate to **System -> IP Network -> Nodes ID: Server Media Cards**. The Node ID Telephone page will appear as shown in figure 13.
- Then click on the **Node ID # 555.**

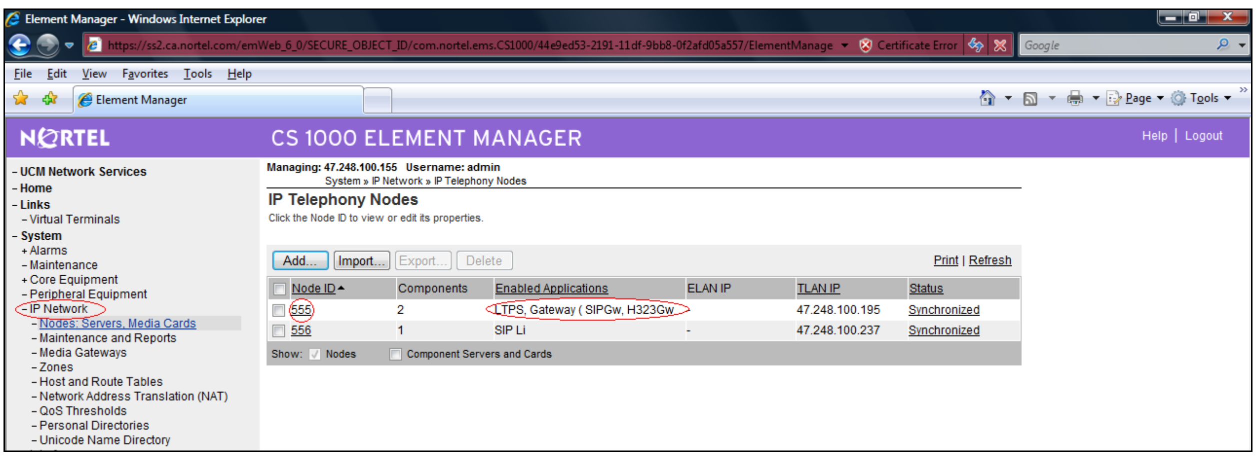

Figure 13: IP Telephone Nodes page of EM

- The **Node Details** page of Node **555** page is displayed as shown in Figure 14.

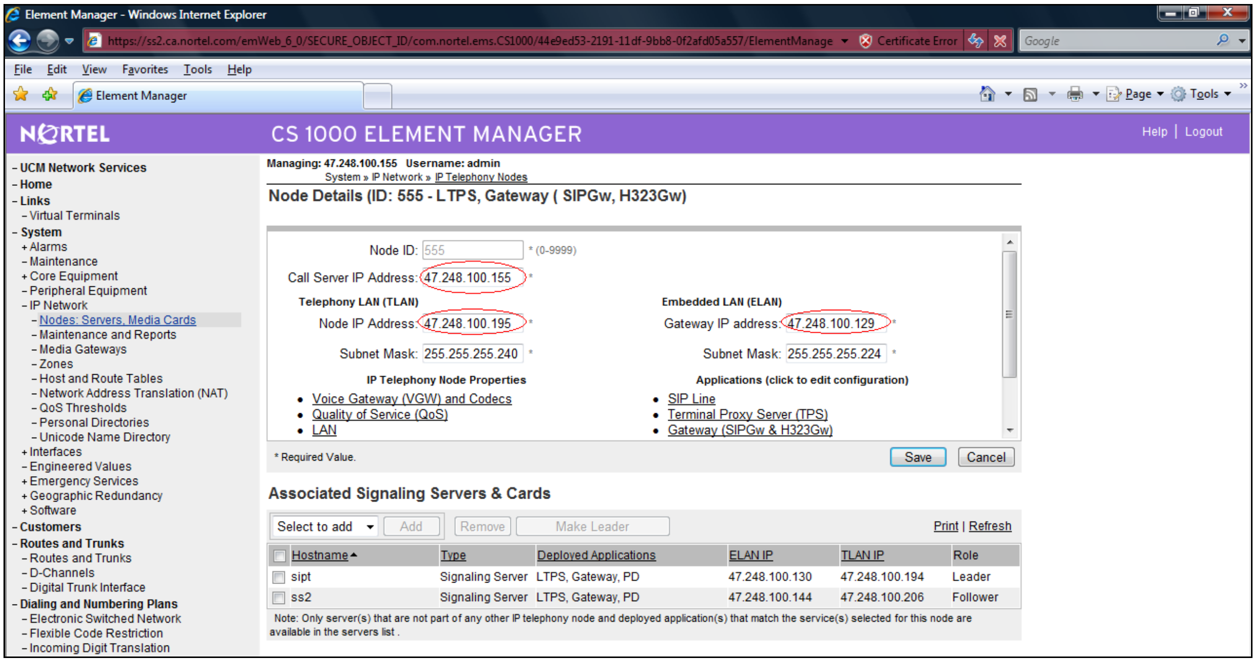

Figure 14: Node ID 555 page on the EM

- From the **Node Details** page of the Node ID 555, under **Applications** attribute, click on the **Gateway (SIPGw & H323Gw)** link and the **Virtual Trunk Gateway Configuration Details** is displayed as shown in Figure 15. The **General** and **SIP Gateway Settings** sections need to be filled in as shown in red circle in Figure 15. Other are left as default values.
  <u>**Notes**</u>: **SIP Domain name**: <u>**nortel-dplab.com**</u> → this is the SIP service domain name on the NRS.
  Gateway endpoint name: **nodea** → this endpoint name has to be matched with the endpoint name of CS 1000 Node ID **555** on the NRS.

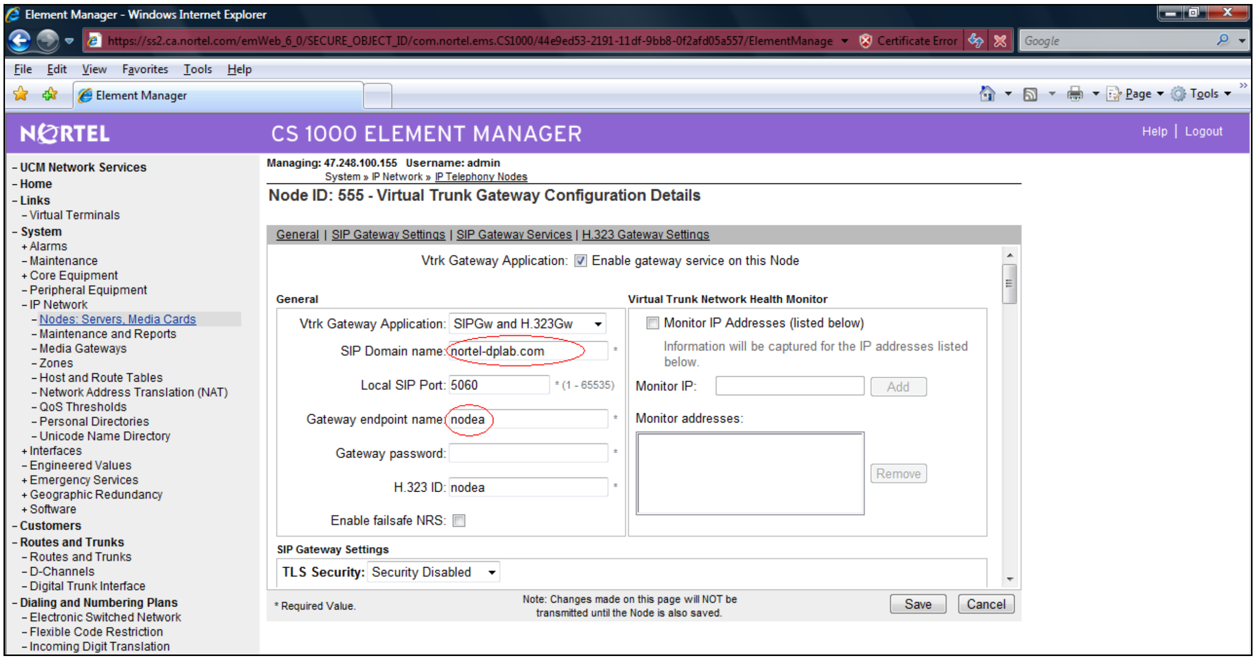

Figure 15: Virtual Trunk Gateway Configuration Details Page of EM

Under **the SIP Gateway Settings** section, the details configuration is filled out as shown in red circles, Figure 15b. Others are at default values.

Notes:

Primary TLAN IP Address: 47.248.100.234 → this is the primary IP address of the NRS

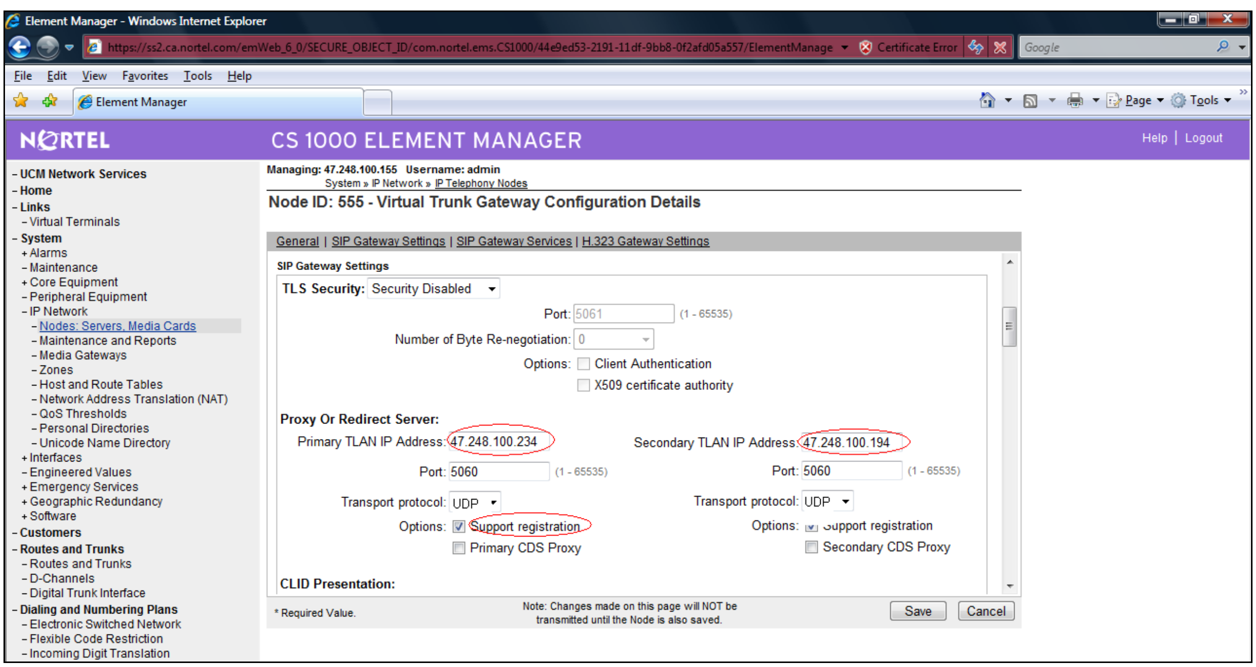

Figure 15b: Virtual Trunk Gateway Configuration Details page of EM

On the same page as shown in Figure 15b, scroll down the parameters box to the **SIP URI Map** section.
Under the **Public E.164 Domain Names**, for
- **Special Number**: leave this SIP URI field as blank
- **Unknown**: leave this SIP URI field as blank
- **Vacant number**: leave this SIP URI field as blank
- **National**: leave this SIP URI field as blank
The remaining fields can be left at their default values as shown in Figure 15c. Click on the **Save** button.

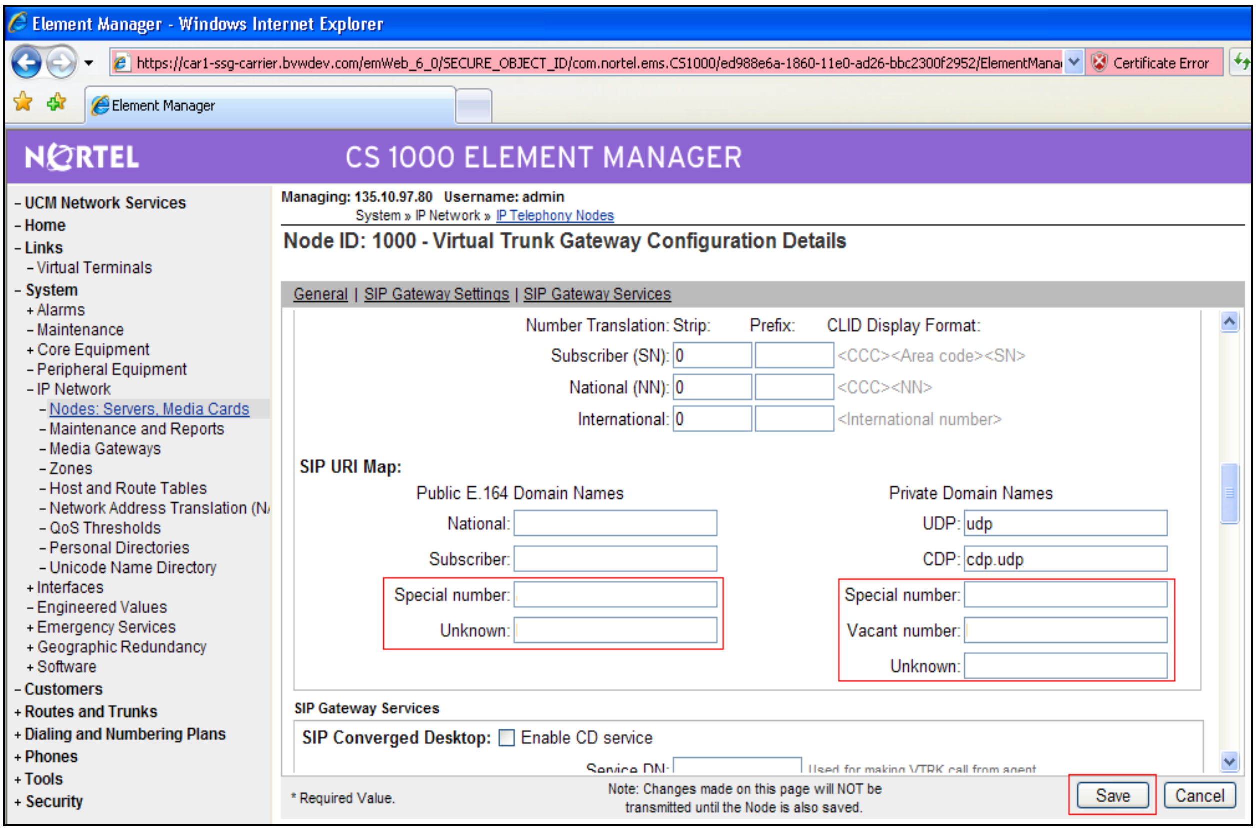

Figure 15c – Virtual Trunk Gateway Configuration Details Page

Note: This will remove the phone context information in the SIP invite URL.

- Click on the **Save** button to save configuration details. When finished, the system will bring back the Node ID page (not shown). Then Click on the **Save** button on the Node ID page and that will take the user to the **Node Saved** page as shown in Figure 16. Click on the **Transfer Now** button as shown in Figure 16 and **Start Sync** button as shown in Figure 17 respectively to complete the changes.

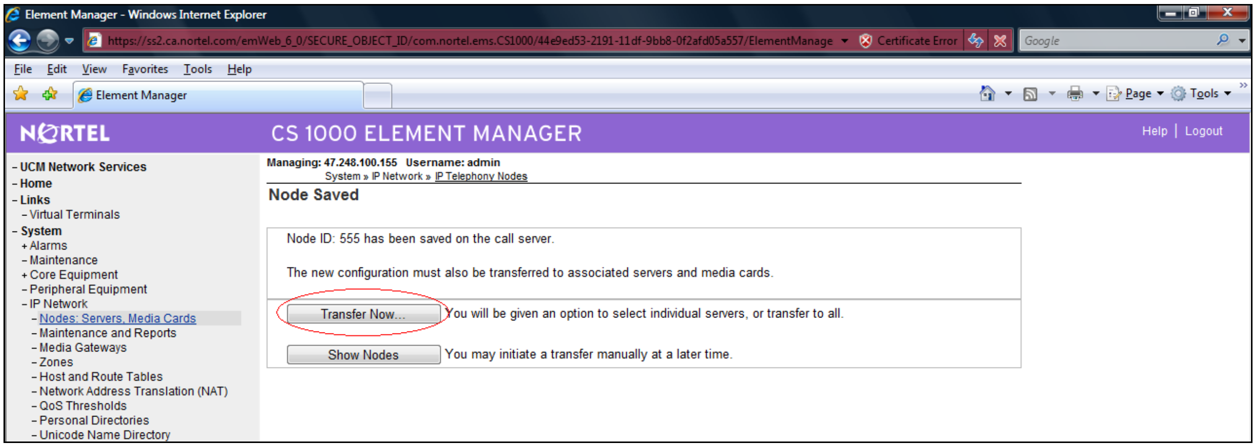

Figure 16: Transfer Now action of the Node Saved Page

Note: When changes on the ***IP Networks***, ***Node: Server, Media Cards*** are done by Element Manager, the process of ***Transfer Node*** and ***Start Sync*** above needs to be applied.

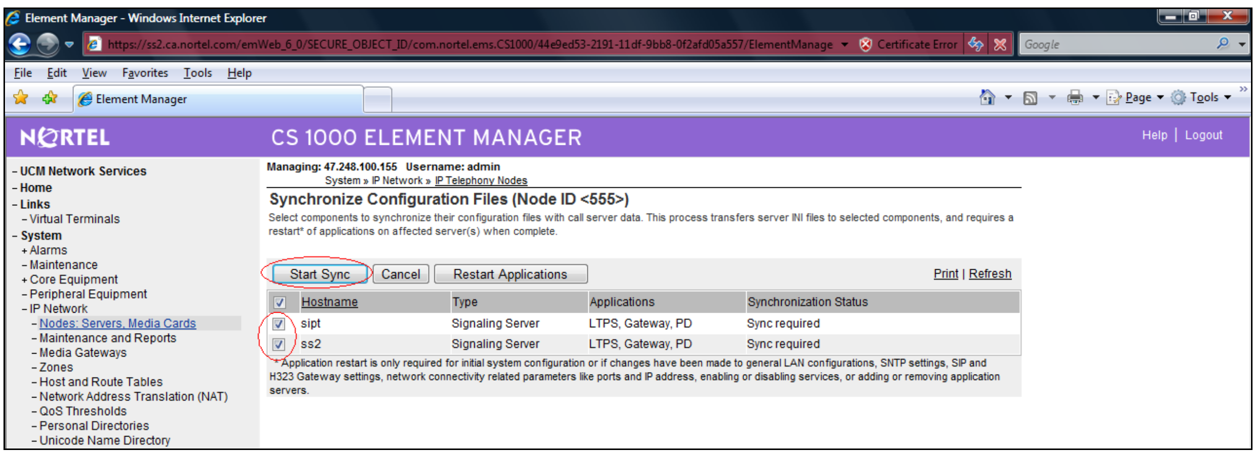

Figure 17: Synchronized Changes on the EM

4.3.2. Creating Route Data Block (RDB) for call server on the EM

To create a new route data block for Node ID ***555*** on the EM, follow the steps below:

- On the left menu column, navigate to the ***Routes and Trunks*** -> ***Routes and Trunks*** menu item, the **Routes and Trunks** details page will appear as shown in Figure 18.
- At the ***Customer 0***, click on the **Add route** button as shown in red circle.

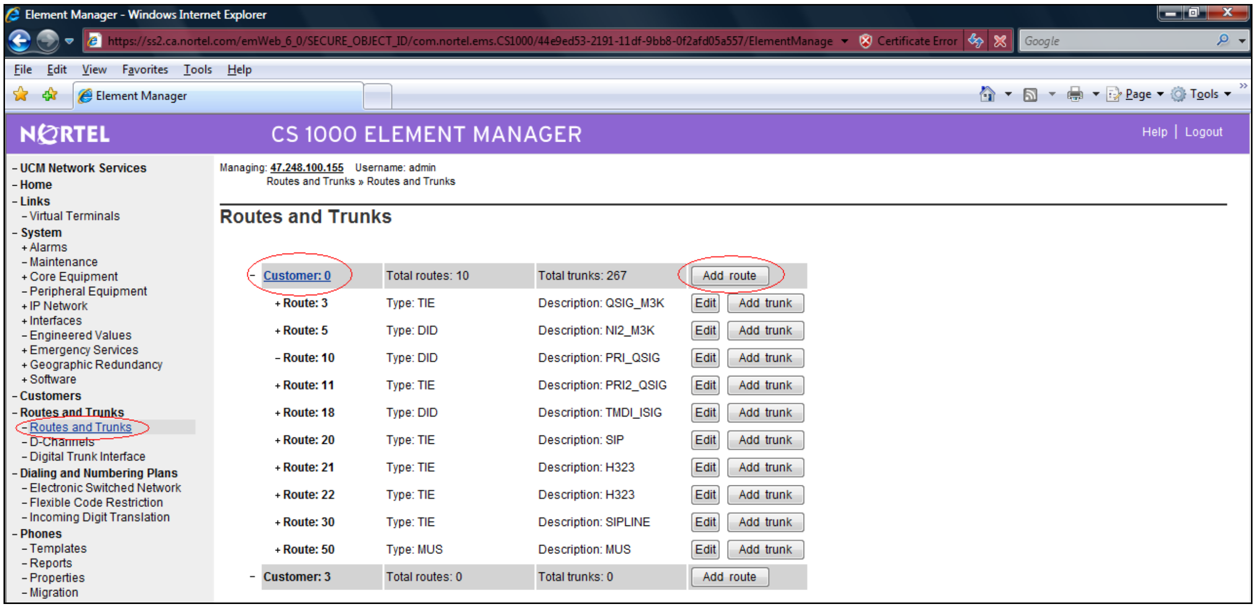

Figure 18: Routes and Trunks webpage

The **New Route Configuration** page for Customer 0 will appear as shown in Figure 19. Under **Basic Configuration** section, the attribute; **Route data block (RDB) (TYPE)**, **Customer number** and **Trunk Type M911P** are left at default values. The others fields are being populated as shown in red circle.

When the **Integrated services digital network option (ISDN**) checkbox is checked, the details of the ISDN service options are displayed (not shown). Fill out all the attributes with the details below:

- **Mode of operation (MODE):** ***ISLD***
- **D channel number (DCH):** ***20*** **→ this channel has to be created. Refer to [3]**
- **Interface type for route (IFC):** ***SL1***
- **Private network identifier:** ***0001***
- **Network calling name allowed (NCNA):** ***Checked***
- **Network call redirection (NCRD):** ***Checked***
- **Channel type (CHTY):** ***B-channel (BCH)***

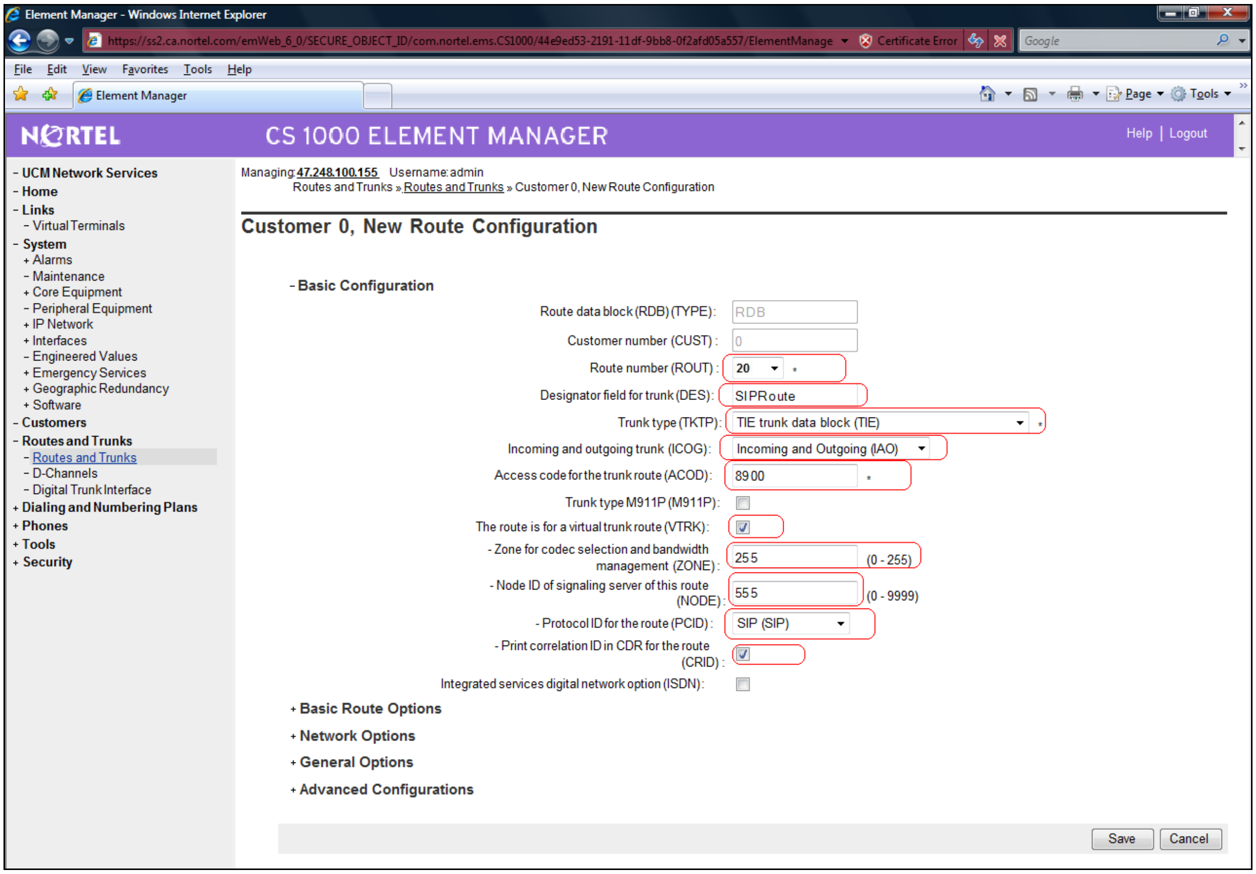

Figure 19: Basic Configuration of RDB

Click on the **Basic Route Options**, the attributes below are populated as shown in red circles, in Figure 20.

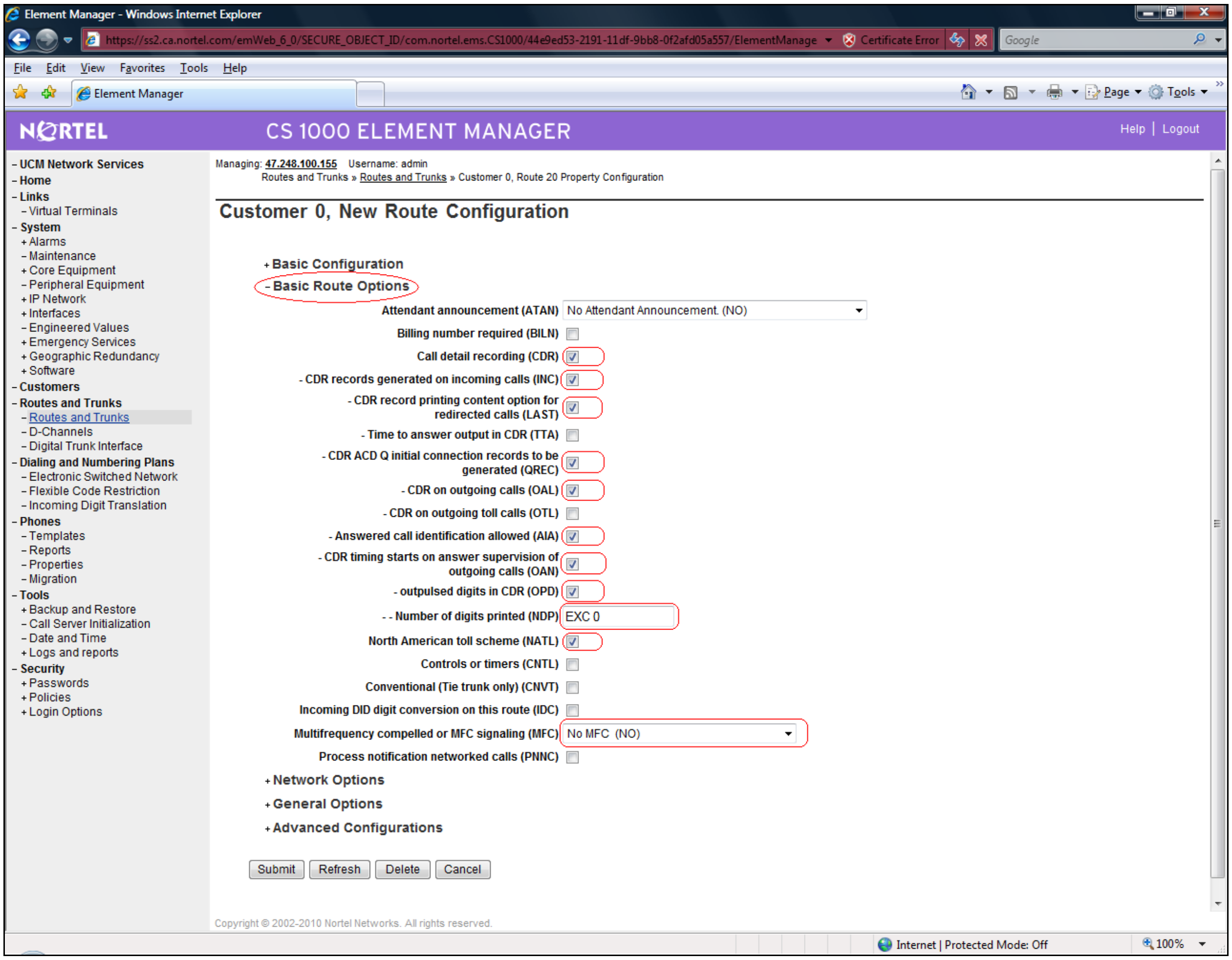

Figure 20: Basic Route Options of RDB

Click on the **Network Options**, the attributes below are populated as shown in red circles, in Figure 21.

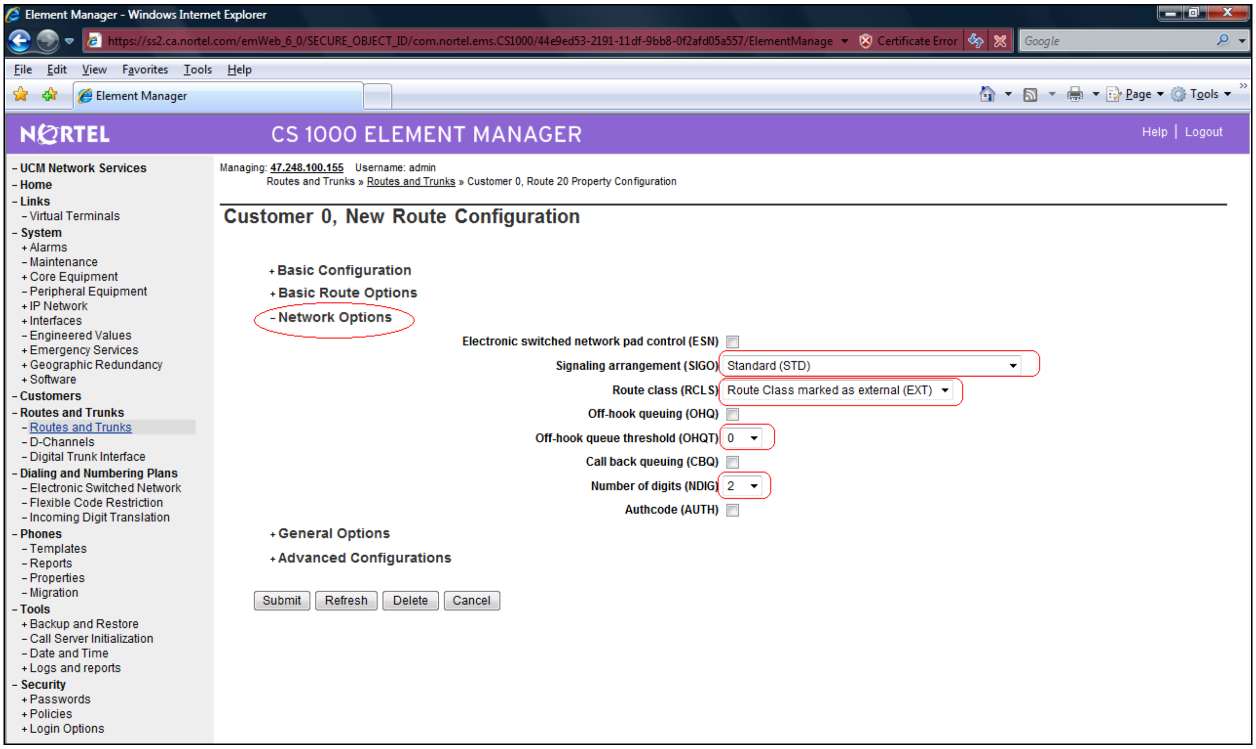

Figure21: Network Options of RDB

Click on the **General Options**, the attributes below are populated as shown in red circles, in Figure 22.

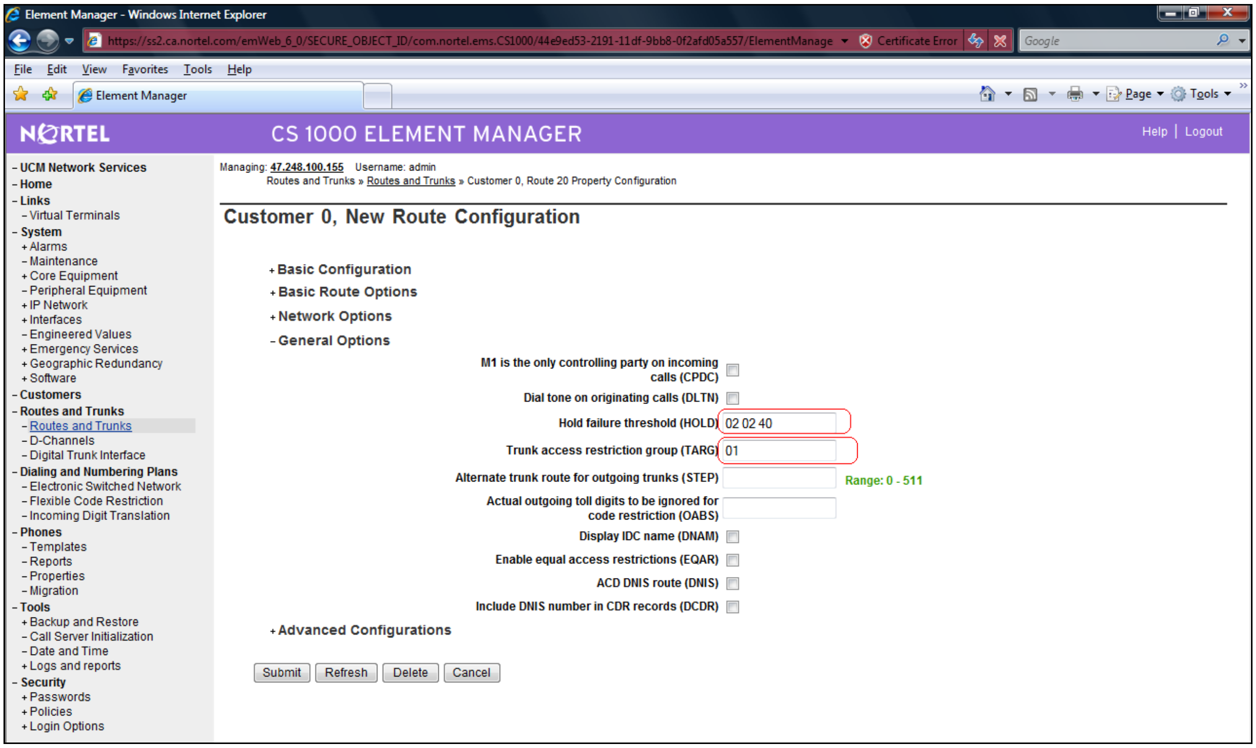

Figure 22: General Options Section of RDB

Click on the **Advanced Configurations**, the attributes below are populated as shown in red circles, in Figure 23 and 23b.

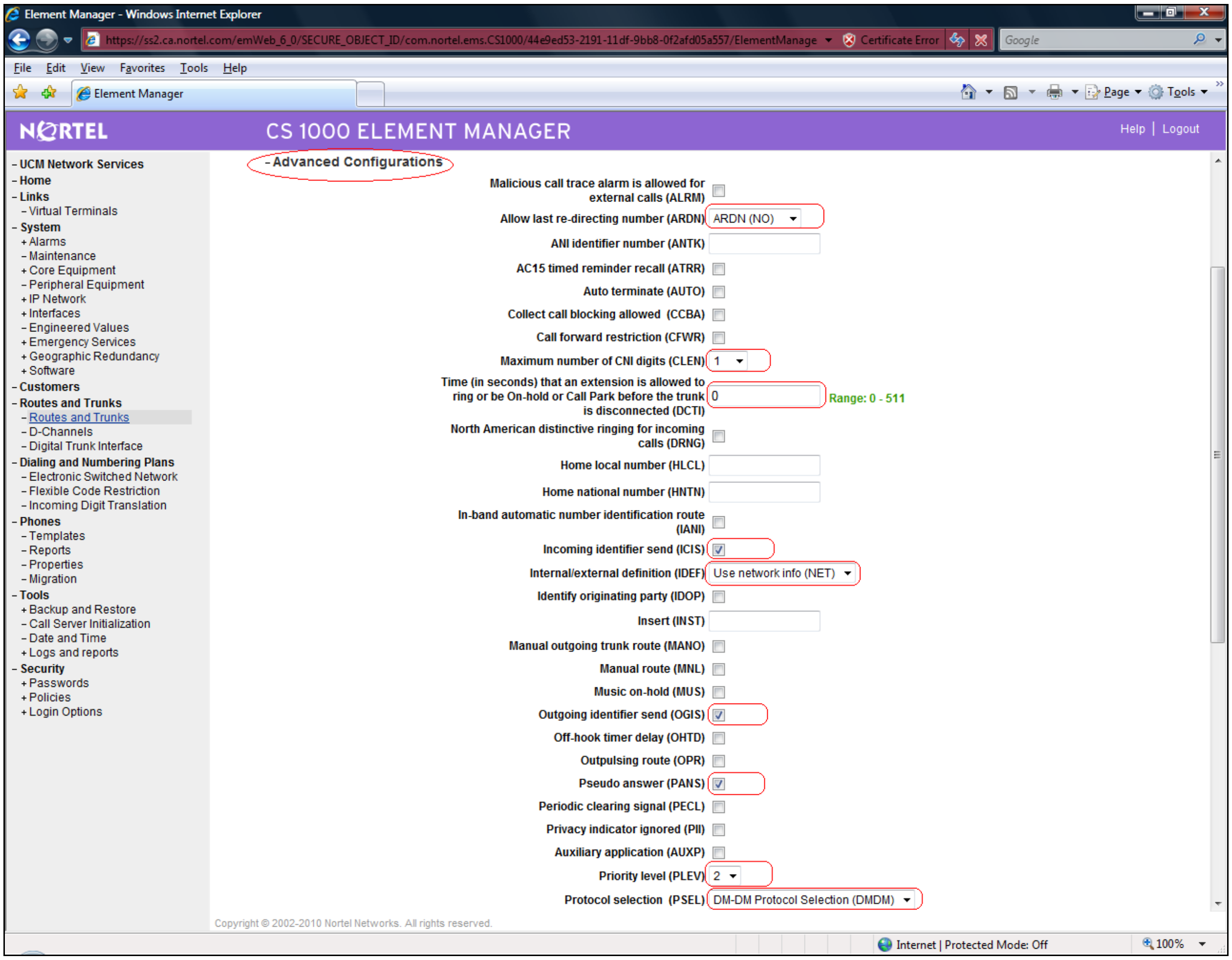

Figure 23: Advanced Configurations Section of RDB

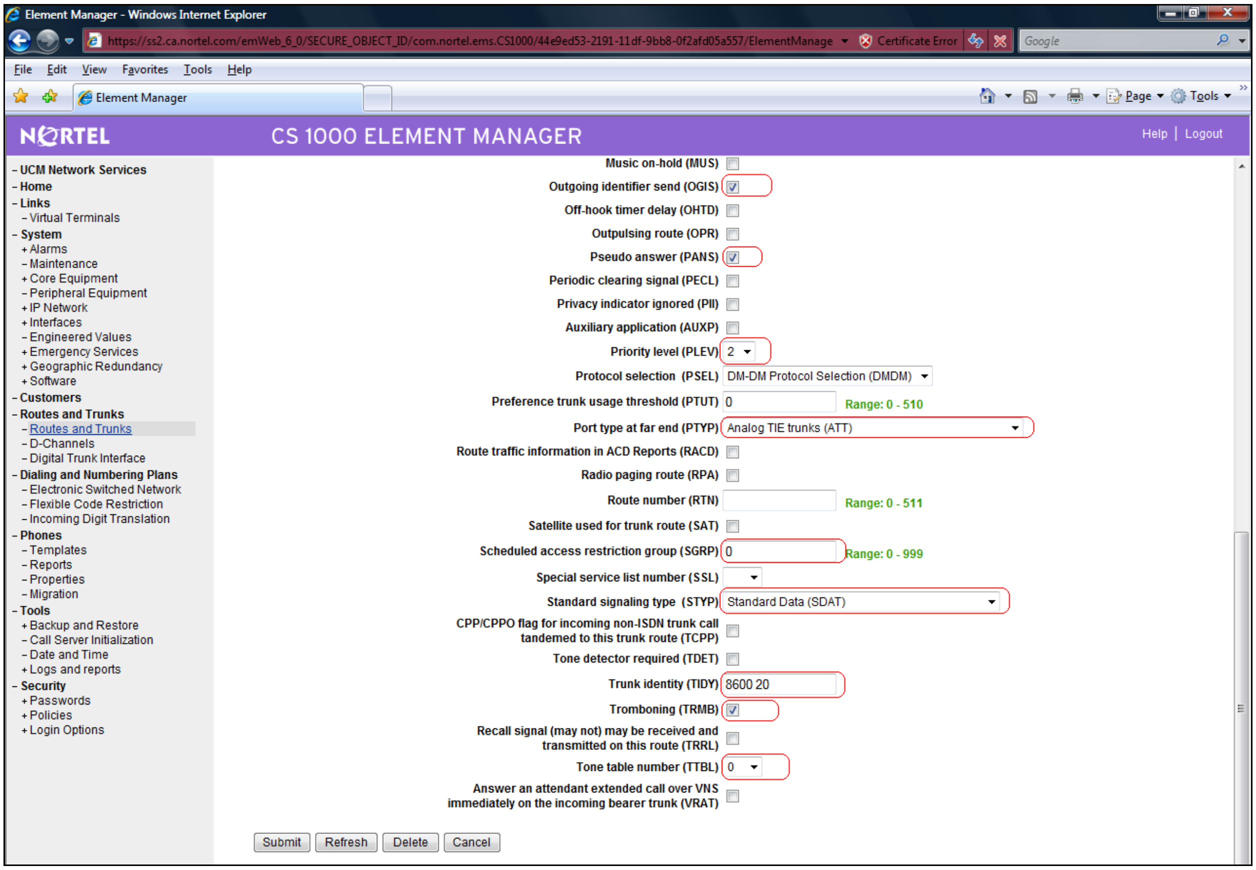

Figure 23b: Advanced Configurations Section of RDB (Continued)

Click on the **Submit** button to save and complete the newly created route 20.

4.3.3. Creating SIP trunks for call server Node ID 555 on the EM

To create SIP Trunks between SIP signaling gateway to the target NRS, on the left menu column, navigate to the ***Routes and Trunks*** -> ***Routes and Trunks*** menu item, the **Routes and Trunks** details page will appear as shown in Figure 24. Click on the **Customer 0** and choose the newly create **Route 20** that SIP trunks belongs to in order to add associated trunk.

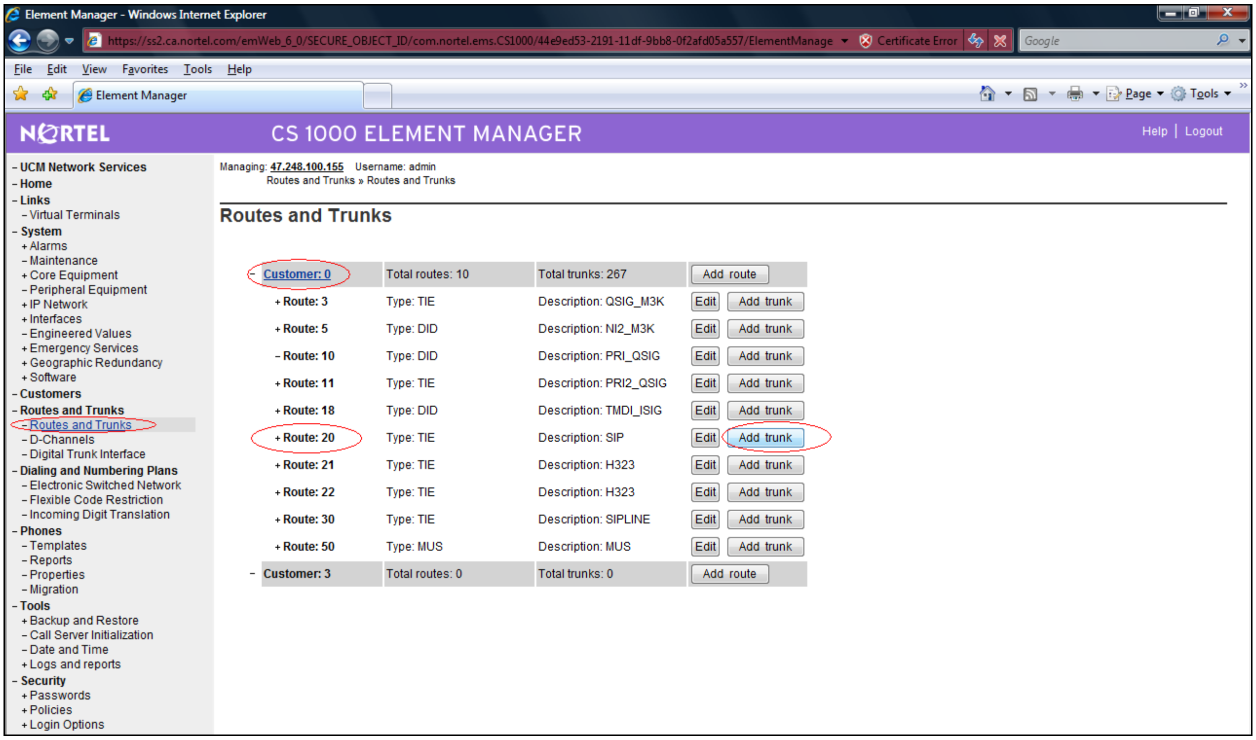

Figure 24: Routes and Trunks Properties Page of the EM

The **New Trunk Configuration** page will appear. Click on the **Basic Configuration**, the attributes below are populated as shown in red circles, in Figure 25.
At the **Class of service (CLS)** attribute, click on the **Edit** button, the list of class of Service options are displayed (not shown). Click on the drop down list of the **Restriction level** and select **Unrestricted (UNR)**.

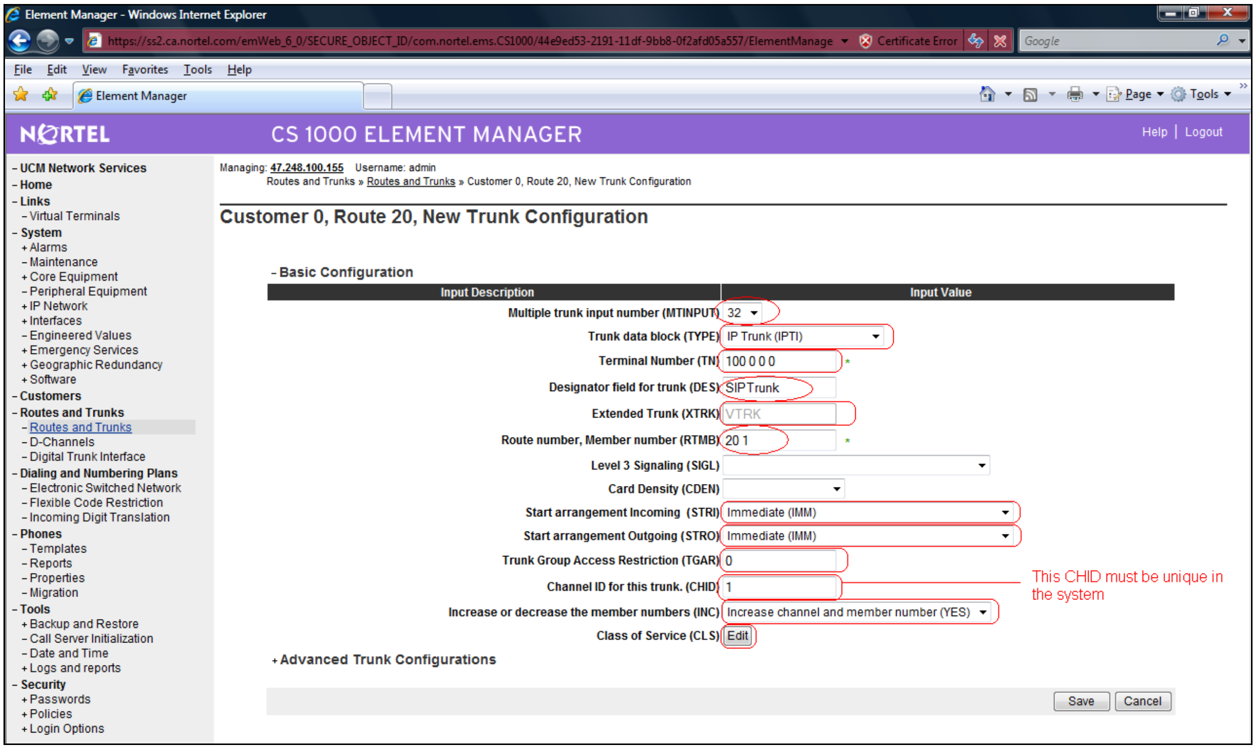

Figure 25: New Trunk Configuration

Click on the **Advanced Trunk Configurations**, the attributes below are populated as shown in red circles, in Figure 26.

Click on the **Save** button, in order to complete the newly created Trunks.

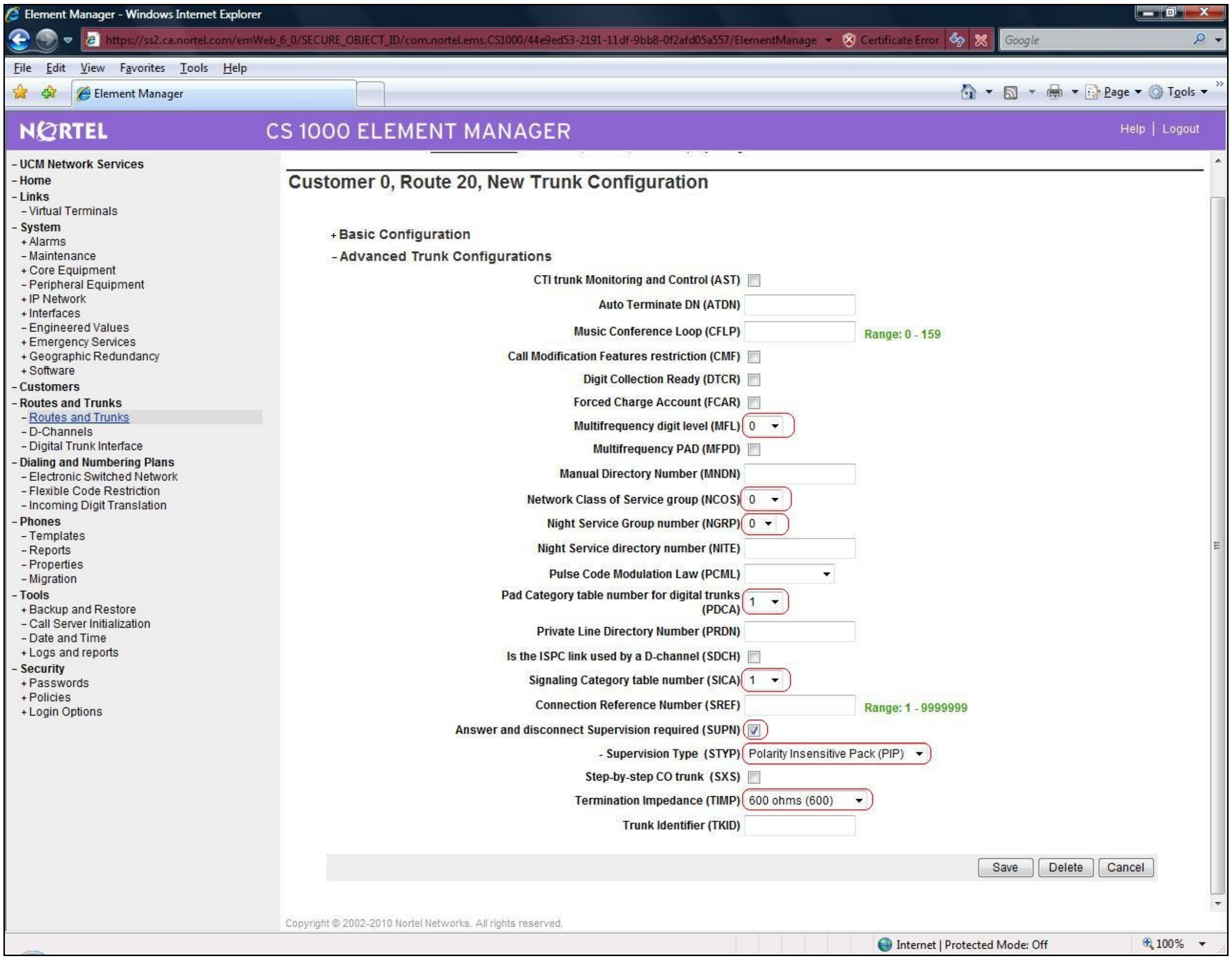

Figure 26: Advanced Trunk Configurations

4.3.4. Creating Digit Manipulation Index (DMI) for CS100 on the EM

To create Digit Manipulation Index (DMI), on the left menu column, navigate to the **Dialing and Numbering Plans** -> **Electronic Switched Network** menu item, the **Electronic Switched Network (ESN)** page will appear (not shown). Continue to navigate to **Customer 0 -> Network Control and Services -> Digit Manipulation Block**, the ***Digit Manipulation Block list*** will appear as shown in Figure 27.

Click on the menu drop down list to pick a Digit Manipulation Block Index (7 in this example)

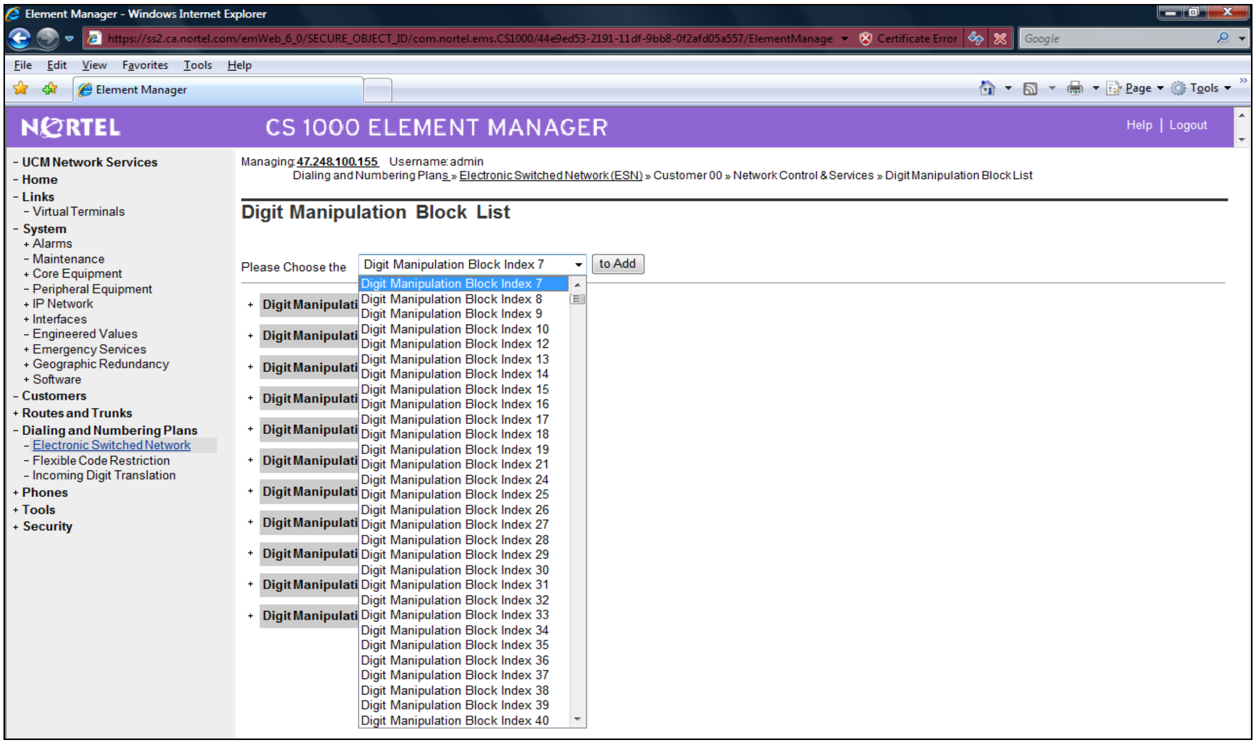

Figure 27: Digit Manipulation Block list

Then click on the **to Add** button, the **Digit Manipulation Block** page will appear as shown in figure 28. Fill in the **Number of leading digits to be deleted (DEL**) with value of **0.**
Note: The dialing plan being used will determine how many digits can be deleted.
Click on the **Save** button, in order to complete the creation of the DMI index.

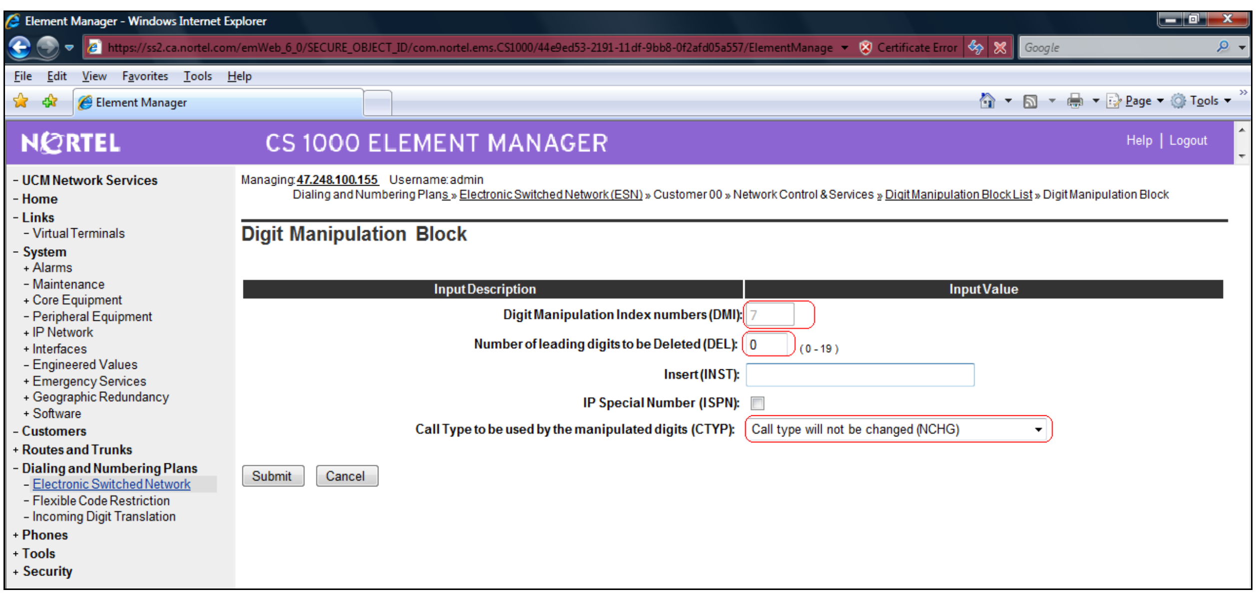

Figure 28: Digit Manipulation Block page

4.3.5. Creating Route List Block (RLB) for call server on the EM

Perform the following steps to create Route List Block (RLB) on the EM.
In the EM, on the left menu column, navigate to the ***Dialing and Numbering Plans*** -> ***Electronic Switched Network*** menu item, the **Electronic Switched Network (ESN)** page will appear (not shown). Continue to navigate to **Customer 0 -> Network Control and Services -> Route List Blocks (RLB),** the **Route List Blocks** details page will appear as shown in Figure 29.
Enter a RLB number to be added in the text box and click on the **to Add** button.

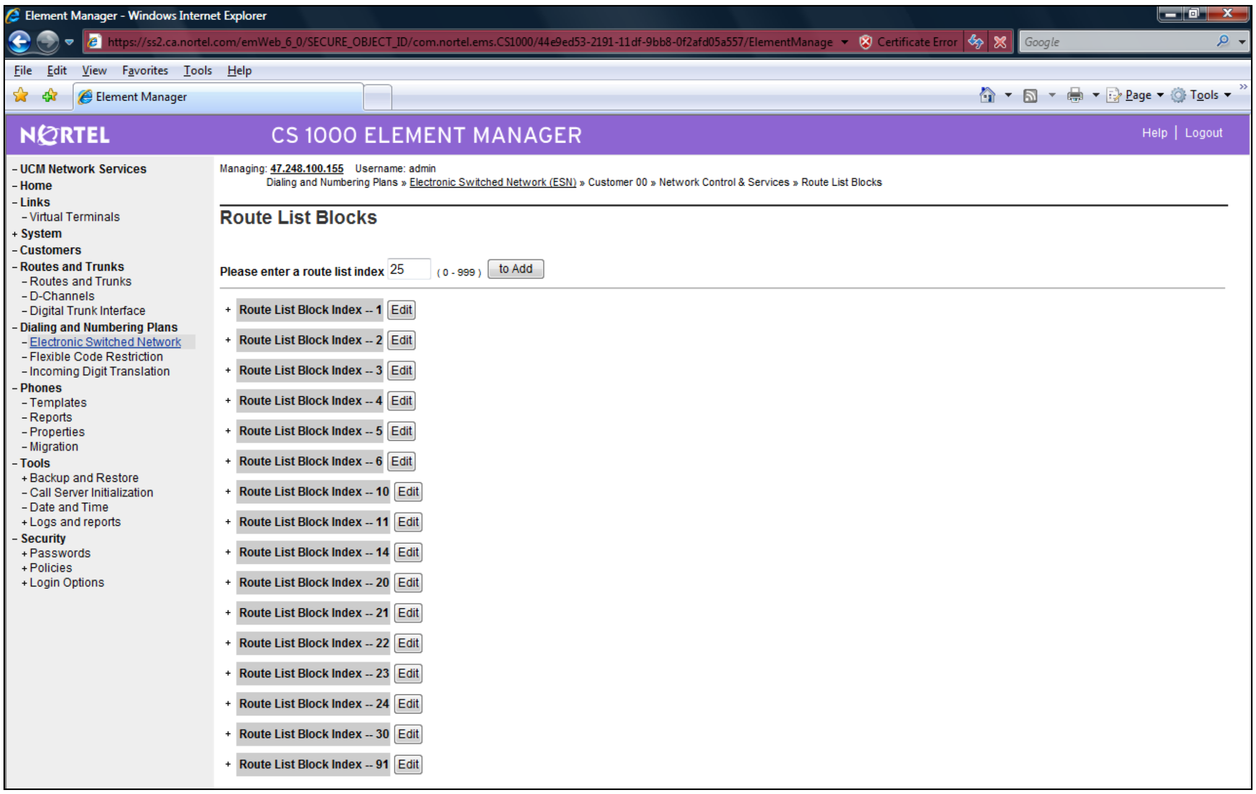

Figure 29: Route List Blocks page

The Route List Block details page for index 25 will appear. Populate the fields, **Route number (ROUT)** and **Digit Manipulation Index (DMI),** with values 20 and 7 respectively as shown in Figure 30. Others are left at default values.

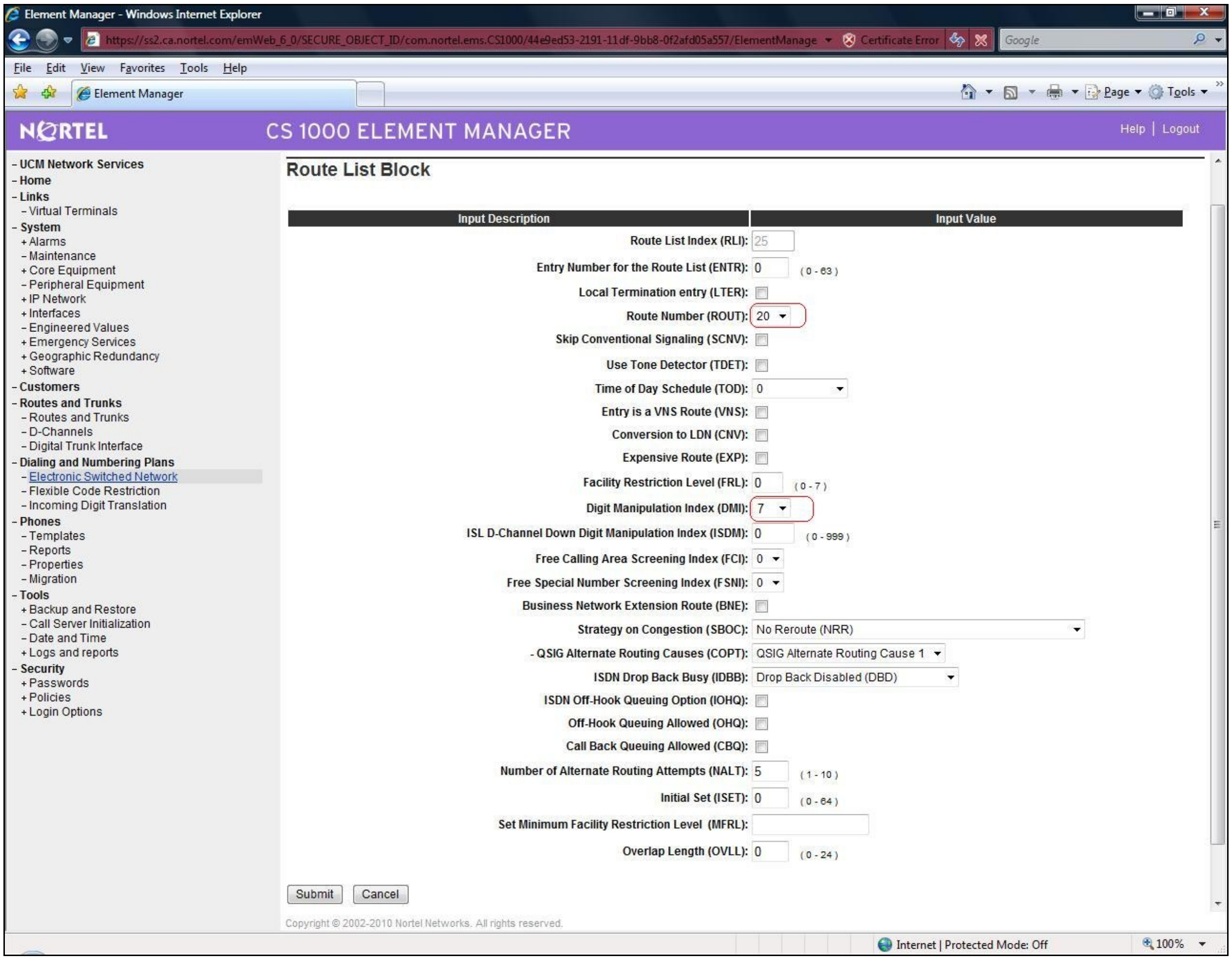

Figure 30: Detail of Route List Block page

4.3.6. Creating Coordinated Dialing Plan (CDP) on the EM

Perform the following steps to create CDP dialing plan on the EM.
In the EM, on the left menu column, navigate to the ***Dialing and Numbering Plans* ->** ***Electronic Switched Network*** menu item, the **Electronic Switched Network (ESN)** page will appear (not shown). Continue to navigate to **Customer 0 -> Coordinated Dialing Plan** -> **Distant Steering Code (DSC),** the **Distant Steering Code list** details page will appear as shown in Figure 31.
From the drop down list, choose **Add** option menu as shown in Figure 31. Enter distance steering code **3200** (in this example) which is the DN prefix of the Vocera Server that has been configured in section 4.2.2, Figure 9. Then click on the **to Add** button.

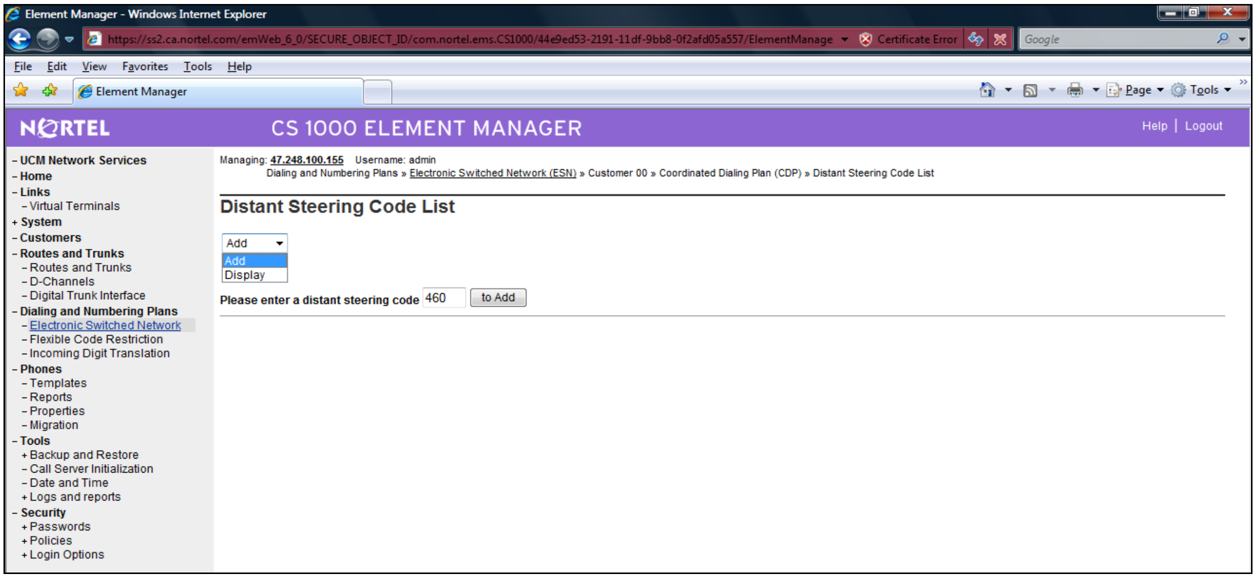

Figure 31: Distant Steering Code List page

The **Distant Steering Code** details page will appear as shown in Figure 32.
Fill in the highlighted red circle attributes as shown in Figure 32.
Other fields are left at default values.
From the **Route List to be accessed for trunk steering code (RLI)** drop down menu, choose the RLI number which has been created, 25, in section 4.3.5, Figure 29.
Click on the **Submit** button to save the changes.

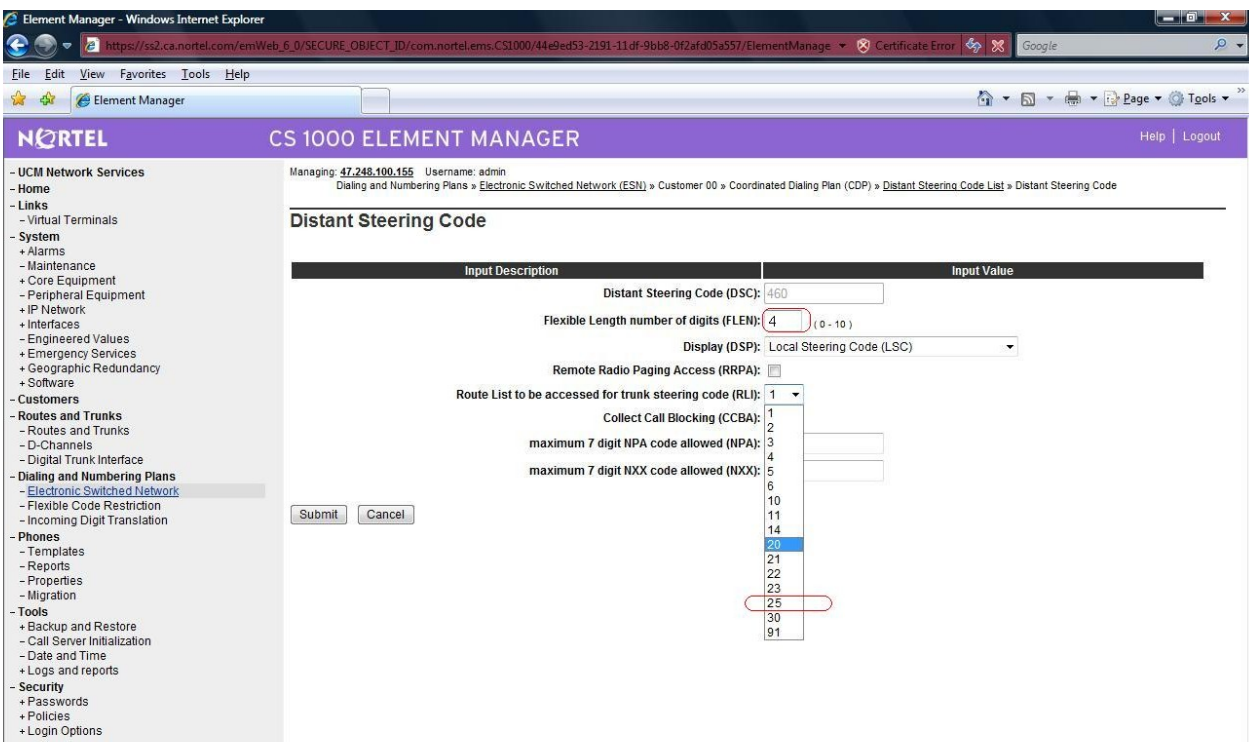

Figure 32: Detail of Distant Steering Code page

5. Configuring Vocera Server

This section describes how to configure the Vocera Communication System to inter-work with the CS 1000 system.

5.1. Configuring Vocera SIP Connectivity to CS1000

Open the Vocera Communication Systems web page by addressing the IP address of the Vocera Server in the Microsoft Internet browser, http://47.248.101.35/. The Welcome to Vocera page will appear (Not shown), then click on the Vocera Administration Console link to get to the console web page as shown in Figure 33.

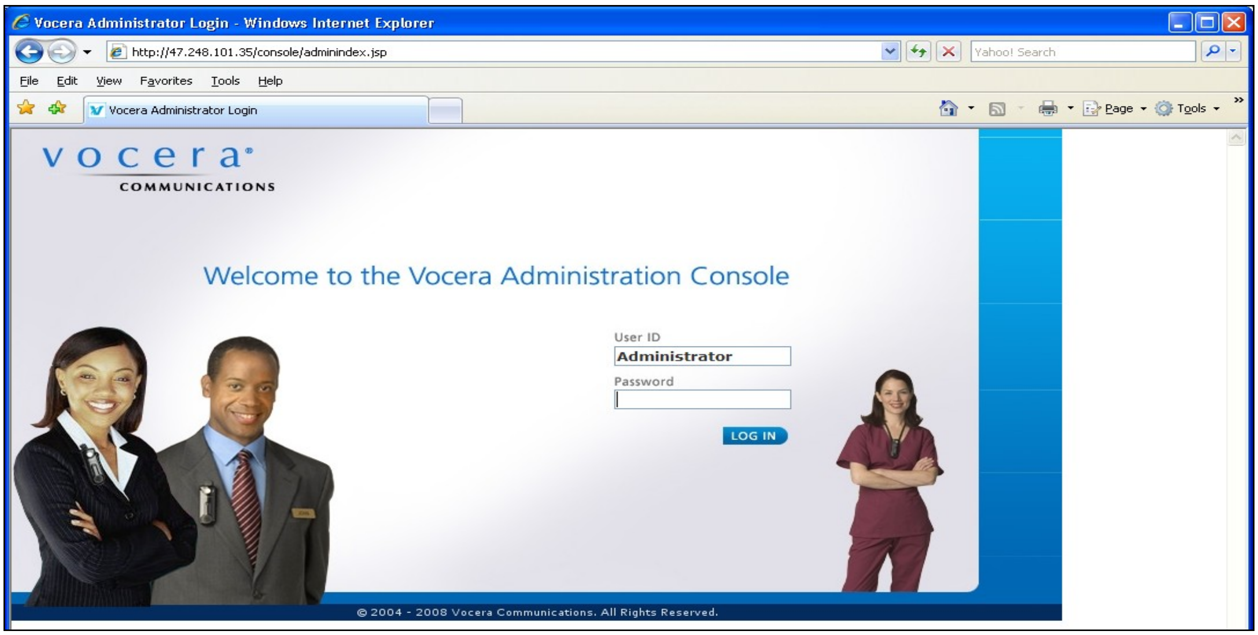

Figure 33: Vocera Administration Console

Input the user name and password to log on the **Console page**, click on the **Log In** button to log in. The screen shown in Figure 34 will appear with the **Status Monitor** menu page as default.

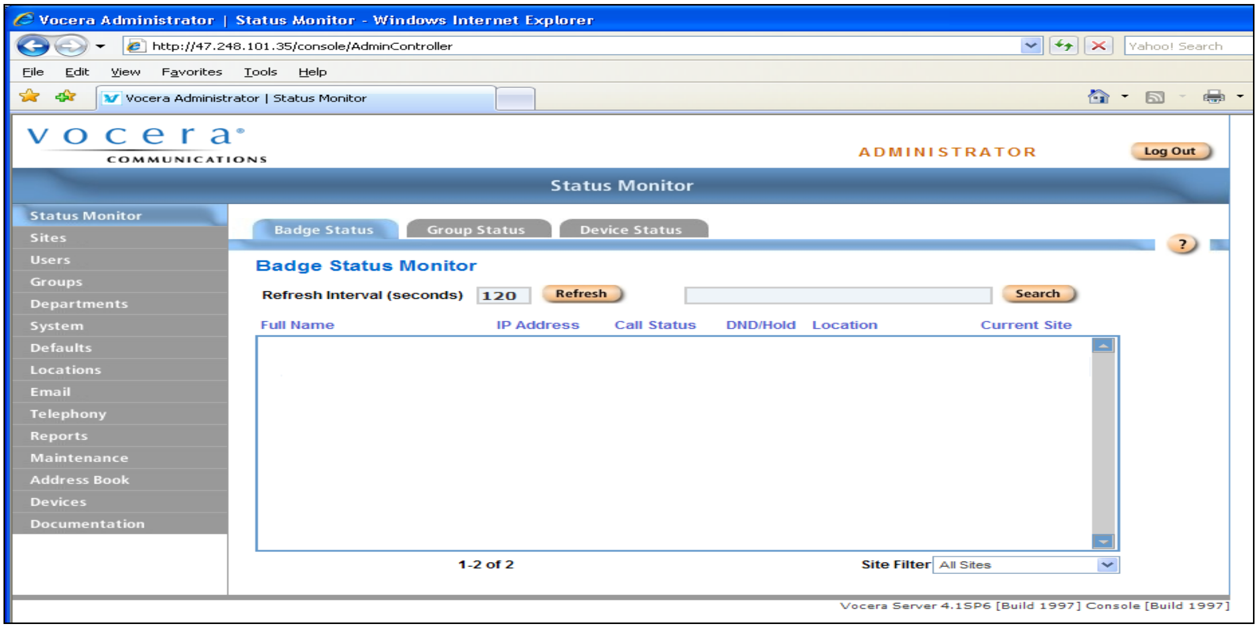

Figure 34: Administrator Console Web Page

For all the details on the configuration of Vocera Communication System, user can click on the **Documentation** option on the left column menu. In the Administration and Configuration column, select on the Telephony Configuration Guide to view the details description of all the available attribute settings.

To configure the Vocera Server to work with the CS1000, click on the **Telephony** option on the left menu column. The **Telephony** page will appear with the **Basic Info** menu tab being selected as default, as shown in Figure 35. Fill in the details of the highlighted attributes in the red ovals, for the CS1000, the **Call Signaling Address** should be the Full Qualified Domain Name of the CS1000, not the IP address. Others fields are optional. Then Click on the **Save Changes** button.

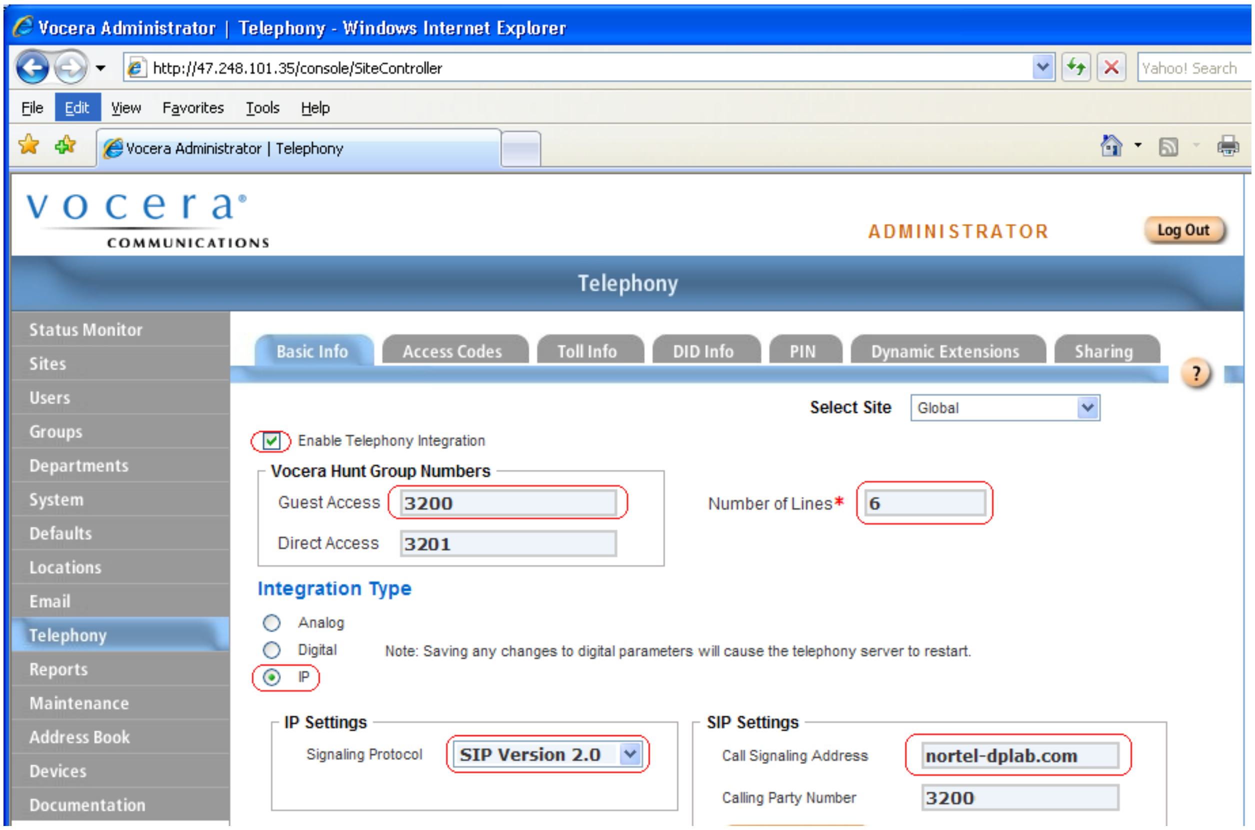

Figure 35: Telephony Configuration

To configure the dialing rule on the Vocera Server, navigate to the **Access Codes** tab, fill in the red highlighted text box of the attributes as shown in the Figure 36. Then click on the **Save Changes** button.

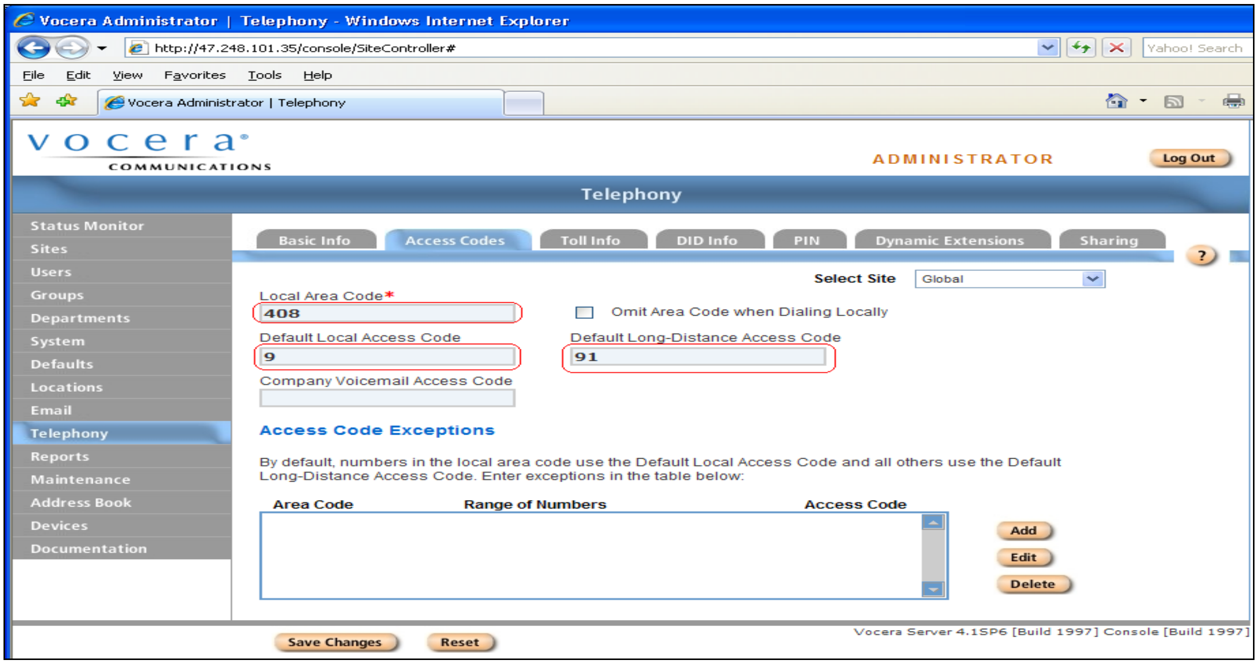

Figure 36: Access Codes Configuration

5.2. Configure Users on the Vocera Server

To configure the users on the Vocera Server to be able to send to and receive calls from the CS1000, as shownin Figure 36 above, click on the **Users** menu option. The Users page will appear as shown in Figure 37.

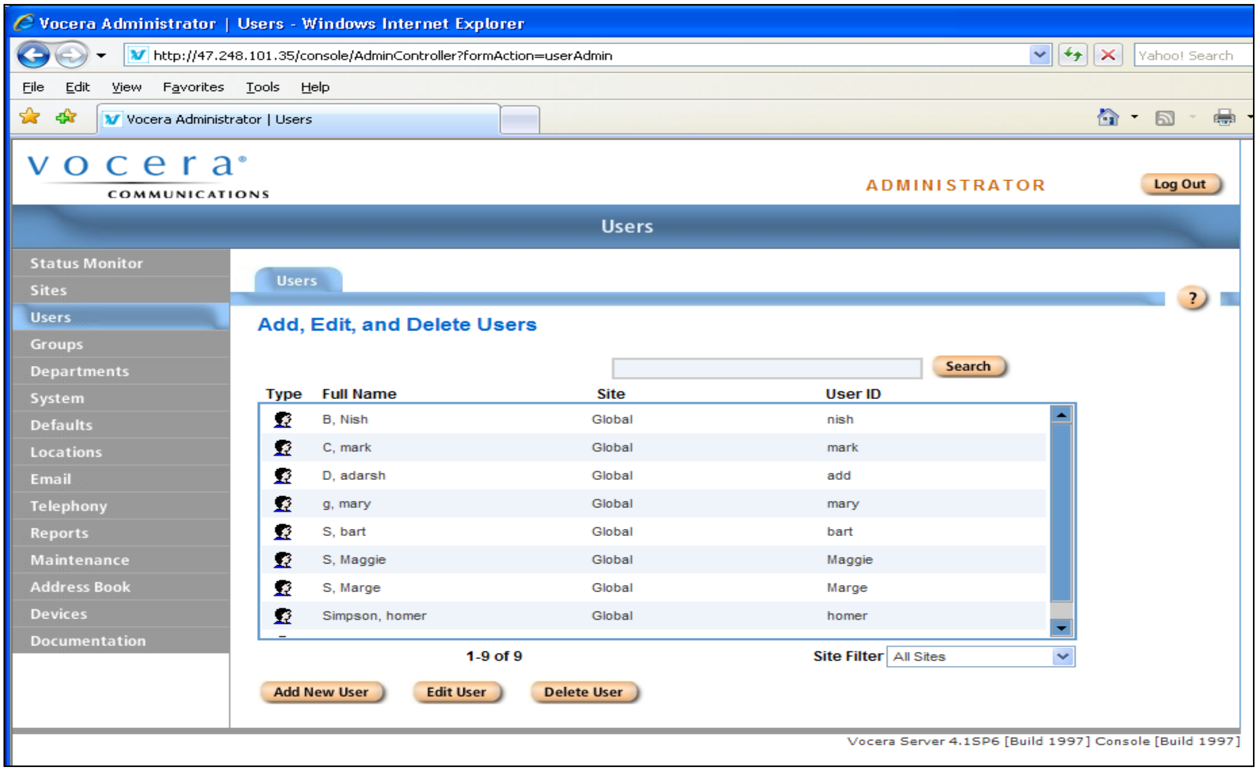

Figure 37: User Page

To add a user, click on the **Add New User** button, the user **Info** detail configuration page will appear as shown in Figure 38. Fill in the required fields, which are indicated with the red stars. The **Badge ID** field is also required to be filled in with a valid badge number. Others are optional.

Click on the **Save** button, to save the configuration details.

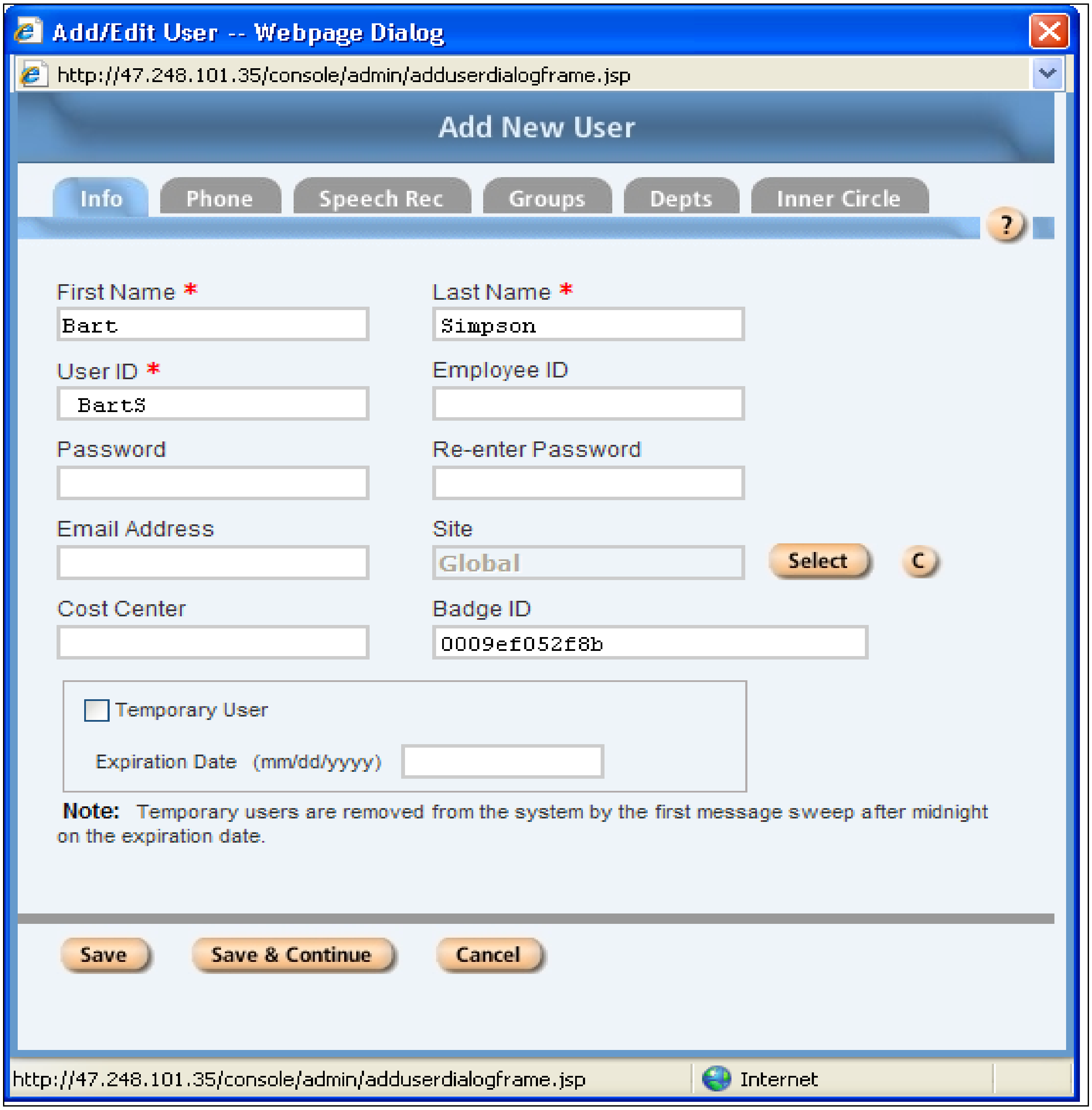

Figure 38: System VOIP Configuration

From the **Add New User** page, click on the **Phone** tab to configure user specific phone number information such as **Desk phone or Extension, Home phone,** as shown in Figure 39. Others fields are optional. Click the **Save** button.

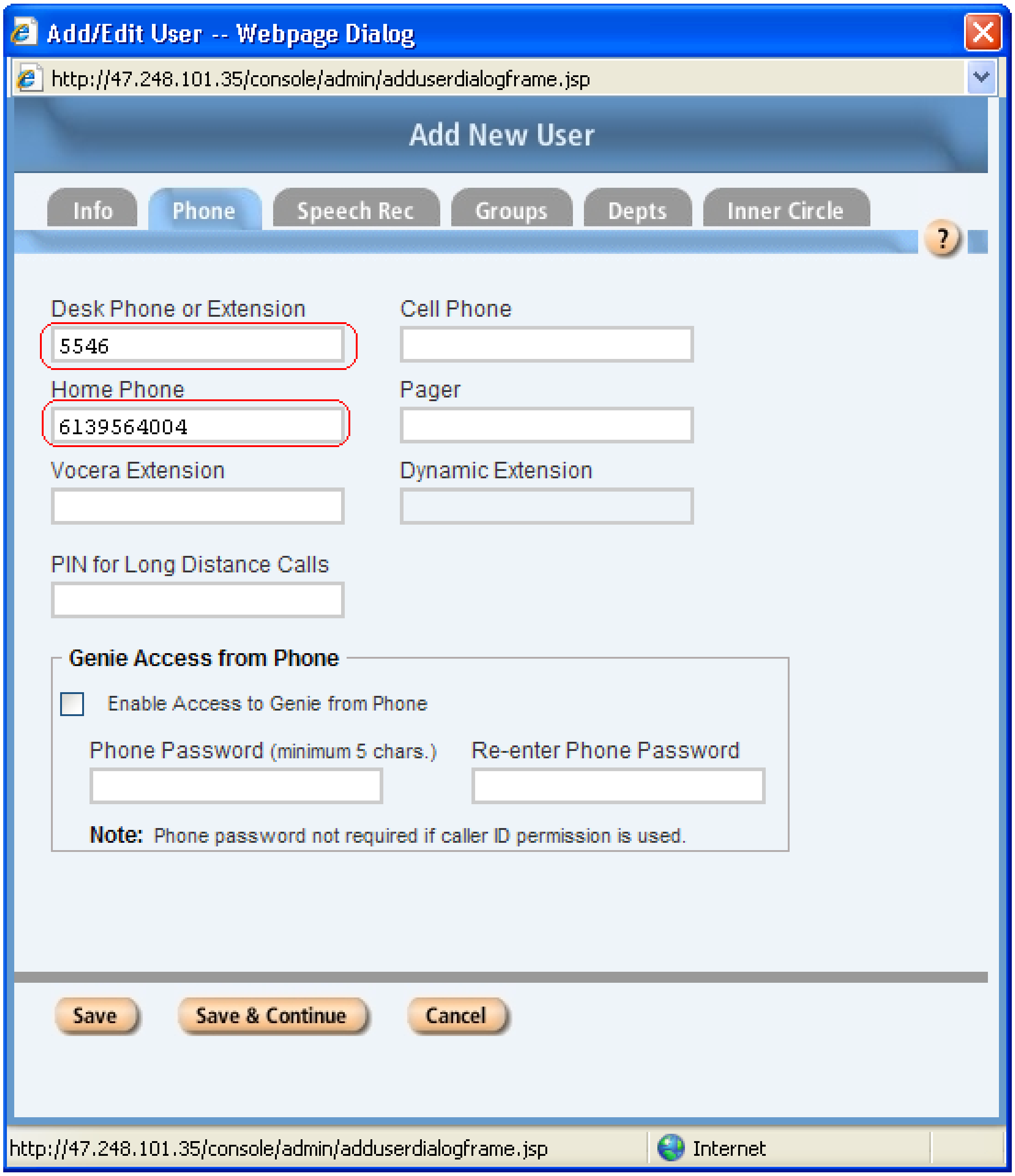

Figure 39: Phone Configuration

Click on the **Group** tab to assign a newly create user to a group with specific permission to use other call features on the Vocera Server. By default, in this example, every new user is assigned to the **Group** Everyone and belonged to the **Site** Global (not shown).

For detail configuration on how these **Groups** and **Sites** are configured, please refer to the **Administration Guide** by clicking on the **Documentation** option menu as shown in Figure 34, under the **Administration and Configuration**.

6. General Test Approach and Test Results

The focus of interoperability compliance testing was primarily to verify the call establishment between the Vocera Communication System and the CS 1000 via SIP trunk.

6.1. General Test Approach

The general test approach was to have different telephone types of the CS 1000 to place a call to the Vocera Server and follow its voice instructions to verify other features of the Vocera Communication System. The main objectives were to verify the Vocera Server successfully performed the following:

- SIP Trunk is established successfully between the Vocera Server and the CS1000.
- Basic calls between the Vocera Server and different telephone types of CS1000 (SIP, non-SIP and emulated PSTN telephones).
- DTMF transmission.
- Conference and Transfer calls from different telephone types of the Avaya CS1000 (SIP, non-SIP and emulated PSTN telephones) to the Vocera Server clients (wireless badge B2000) and vice versa.
- Call Forward (All Call, No Answer, and Busy) and Call Forward to voicemail with Message Waiting Indication (MWI) notification.
- Other telephony features: Busy, Hold and Retrieve calls.

6.2. Test Results

The objectives outlined in the section 6.1 were verified and met. All test cases were executed and they all passed.

7. Verification Steps

The following are typical steps to verify the interoperability between the Vocera Server and Avaya CS 1000 system, please also refer to the Figure 1 for more detail.

- Step 1: Place a call from an IP phone of CS 1000 to the Vocera Server by entering the assigned DN number.
- Step 2: A voice greeting from the Vocera Server should be heard on the IP phone 2 telling the caller to speak a full name or an extension of the callee.
- Step 3: When a spoken full name or an extension of the callee is received by the Vocera Server, it will redirect the call to the wireless badge associated with the assigned extension.
- Step 4: The user on the Vocera Server will hear a Ginie voice asking if the user would like to pick up the call. User, then presses the big circular button on the badge, to accept the call.
- Step 5: Verify that there are clear 2-way voice path between Avaya IP phone and the Vocere wireless badge.

8. Conclusion

All of the executed test cases have passed and met the objectives in the section 6.1. The main features of the Vocera Communication System were successfully tested with basic call features of the Avaya CS 1000 system to make sure they are fully and properly interoperated.

9. Additional References

Product documentation for Avaya products may be found at:
http://support.nortel.com/go/main.jsp

[1] *Communication Server 1000 Maintenance, Release 6.0, Revision 03.16, January 2010, Document Number NN43041-700*

[2] *Troubleshooting Guide for Distributors, Release 6.0, Revision 02.02, December 2009, Document Number NN43001-730*

[3] *Communication Server 1000 Installation and Commissioning, Release 6.0, Revision 03.06, February 2010, Document Number NN43041-310*

[4] *Communication Server 1000 Software Upgrades, Revision 03.12, February 2010, Document Number NN43041-458*

[5] *Communication Server 1000 Linux Platform Base and Applications Installation and Commissioning, Revision 03.10, February 09, 2009, Document Number NN43001-315*

[6] *Communication Server 1000 Unified Communications Management Common Services Fundamentals, Revision: 03.04, September 28, 2009, Document Number NN43001-116*

Product information for Vocera Communication System can be found at
http://www.vocera.com/products/resources/documentation.aspx

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.