



## **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:

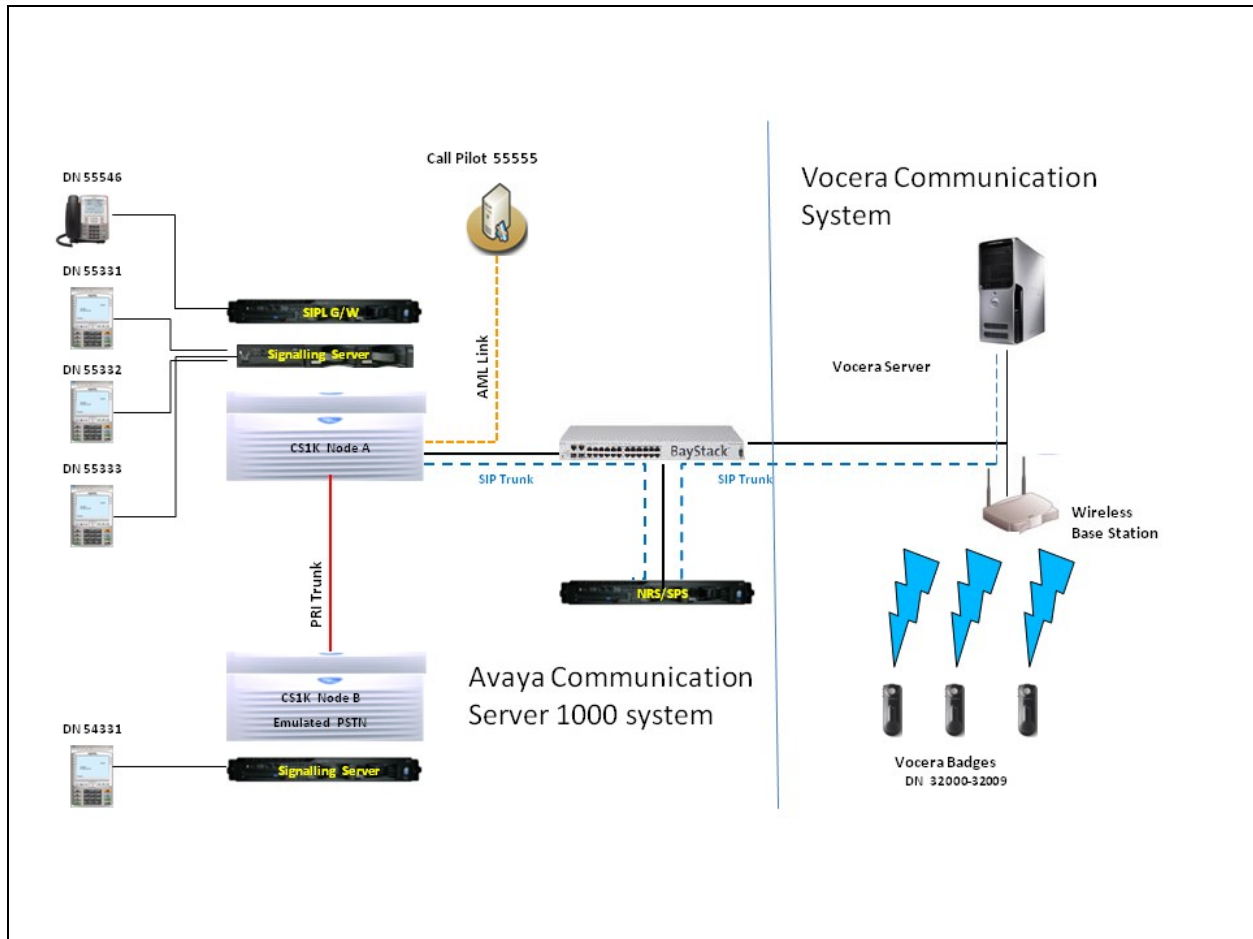
- SIP Trunk is established successfully between the Vocera Server and the CS1000.
- Basic calls between the Vocera Server and different telephone types of CS1000 (SIP, non-SIP and emulated PSTN telephones).
- DTMF transmission.
- Conference and Transfer calls from different telephone types of the Avaya 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](mailto: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	<ul style="list-style-type: none"> <li>Call Server (CPPM): 6.00</li> <li>Signaling Server (CPPM): 6.00.18</li> <li>SIP Line Gateway (HP DL320): 6.00.18</li> </ul>
Avaya CallPilot voicemail system	<ul style="list-style-type: none"> <li>5.0</li> </ul>
Vocera Communication System	<ul style="list-style-type: none"> <li>Vocera Server 4.1SP6</li> <li>Wireless Badge B2000</li> <li>Wireless Base Station</li> </ul>

## 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:
  - o Installed with CS 1000 Release 6.0 Linux Base.
  - o Joined CS 1000 Release 6.0 Security Domain.
  - o Deployed with SIP Trunk Application.

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

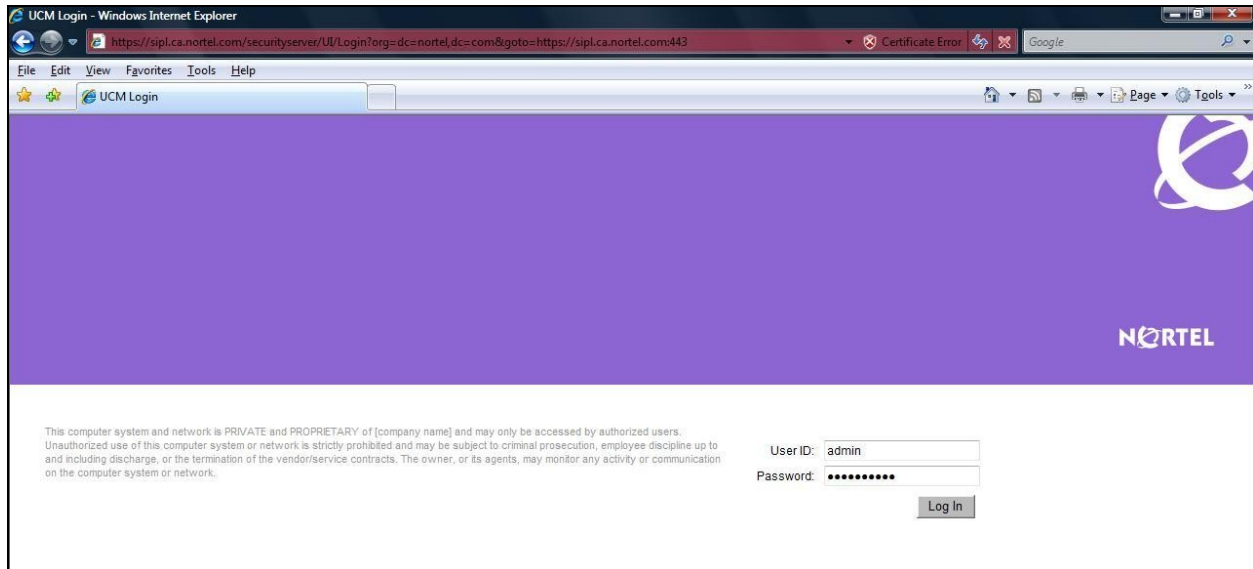
- The following software packages are enabled in the keycode.
- If any of these features have not been enabled, please contact your Avaya account team or Avaya technical support at <http://www.avaya.com>.

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**

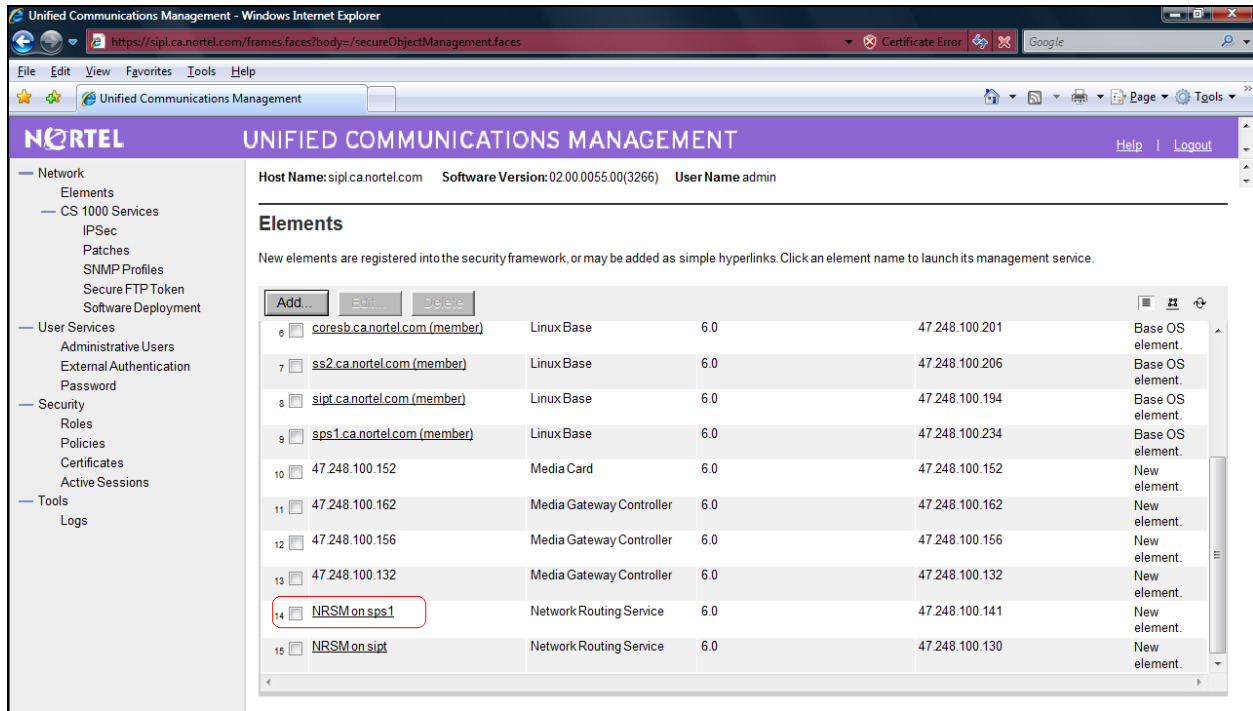
## **4.2. Configuring CS 1000 Network Routing Service (NRS) SIP Proxy Server**

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.

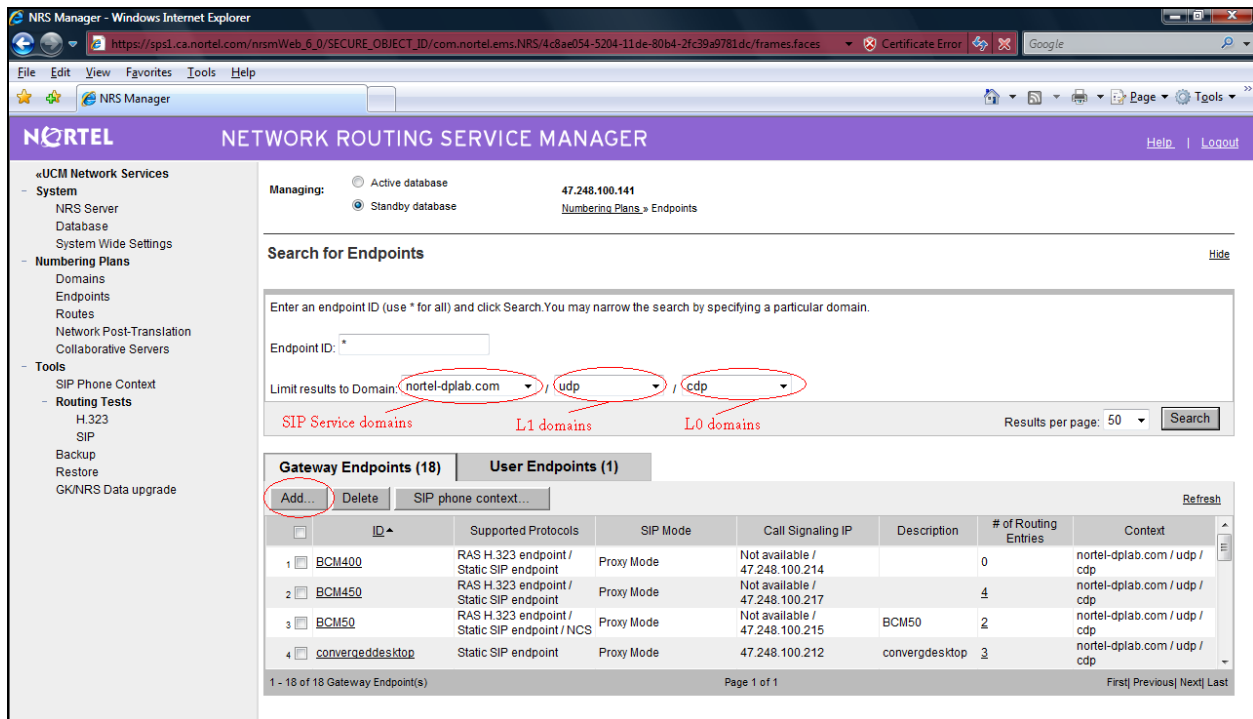


**Figure 3: UCM Navigator Homepage**

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.

**Vocera Manager - Windows Internet Explorer**

https://sps1.ca.nortel.com/nrsmWeb\_6\_0/SECURE\_OBJECT\_ID/com.nortel.ems.NRS/4c8ae054-5204-11de-80b4-2fc39a9781dc/frames.faces

**NORTEL NETWORK ROUTING SERVICE MANAGER**

Managing: ☐ Active database 47.248.100.141  
☒ Standby database

**Edit Gateway Endpoint ( nortel-dplab.com / udp / cdp )**

End point name: Vocera \*

Description: Convergys MS 4.0

TrustNode: ☒

Tandem gateway endpoint name: Not Applicable

Endpoint authentication enabled: Authentication off

Authentication password:

E.164 country code: 1

E.164 area code: 613

E.164 international dialing access code: 9

E.164 international dialing code length: 12 (0-99)

E.164 national dialing access code: 9

E.164 national dialing code length: 10 (0-99)

E.164 local (subscriber) dialing access code:

\* Required value

Save Cancel

**Figure 5: Input Fields of the Vocera Gateway Endpoint**

**NRS Manager - Windows Internet Explorer**

https://sps1.ca.nortel.com/nrsmWeb\_6\_0/SECURE\_OBJECT\_ID/com.nortel.ems.NRS/4c8ae054-5204-11de-80b4-2fc39a9781dc/frames.faces

**NORTEL NETWORK ROUTING SERVICE MANAGER**

Managing: ☐ Active database 47.248.100.141  
☒ Standby database

**Add Gateway Endpoint ( nortel-dplab.com / udp / cdp )**

Static endpoint address type: IP version 4

Static endpoint address:

H.323 support: H.323 not supported

SIP support: Static SIP Endpoint

SIP Mode: ☒ Proxy Mode  
☐ Redirect Mode

SIP TCP transport enabled: ☐

SIP TCP port: 5060

SIP UDP transport enabled: ☒

SIP UDP port: 5060

SIP TLS transport enabled: ☐

SIP TLS port: 5061

Persistent TCP support enabled: ☒

End to end security support: ☐

Network Connection Server enabled: ☒

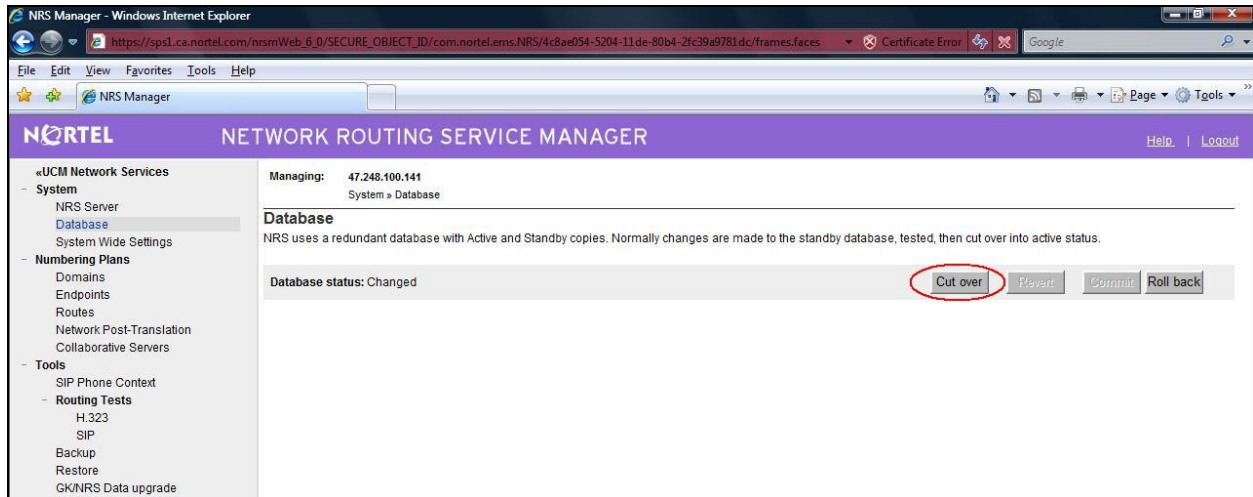
\* Required value

Save Cancel

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

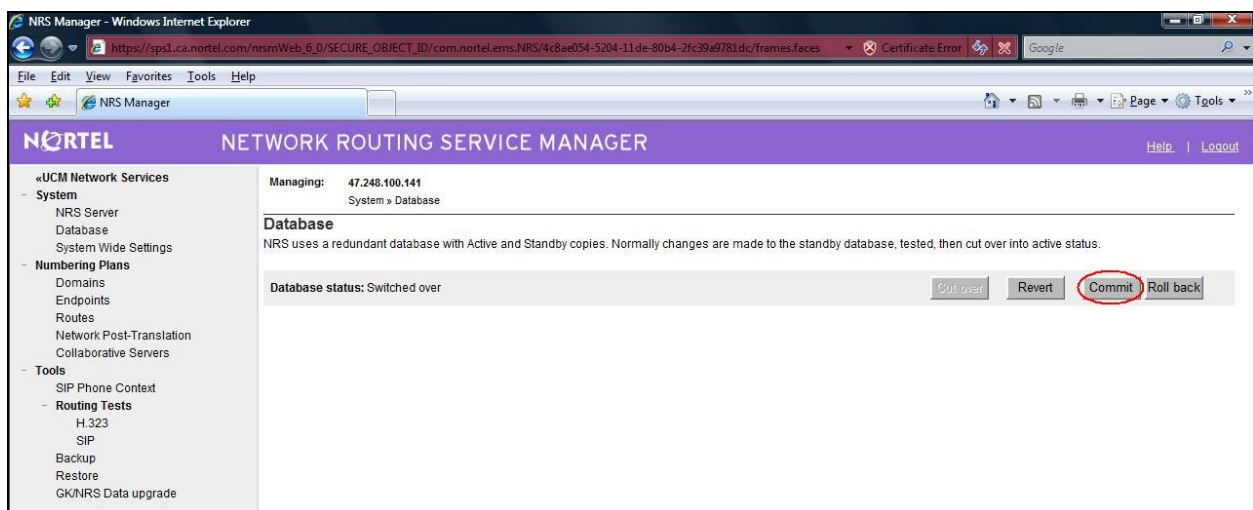


Click on the **Save** button in order to complete the newly created gateway endpoint for Vocera. Select the **Database** on the left column under the **System** menu, the **Database** page will appear as shown in Figure 6. Click on the **Cut over** button to transfer configured data of the gateway endpoint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

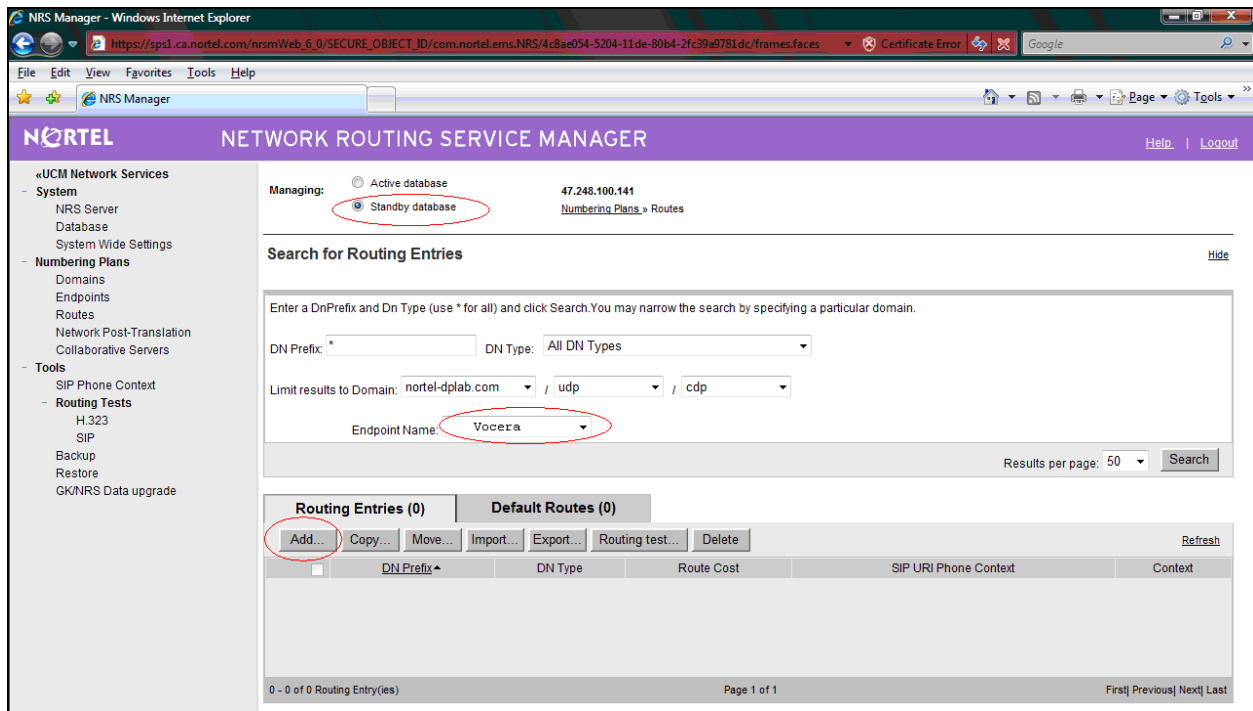
#### 4.2.2. Creating a routing entry for Vocera gateway endpoint on the NRS

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.

NRS Manager - Windows Internet Explorer

https://sp1.ca.nortel.com/nrsmWeb\_6\_0/SECURE\_OBJECT\_ID/com.nortel.ems.NRS/4c8ae054-5204-11de-80b4-2fc39a9781dc/frames.faces

Certificate Error

Google

File Edit View Favorites Tools Help

NRS Manager

Home

Print

Download

Page

Tgols

NORTEL

NETWORK ROUTING SERVICE MANAGER

Help

Logout

«UCM Network Services

System

NRS Server

Database

System Wide Settings

Numbering Plans

Domains

Endpoints

Routes

Network Post-Translation

Collaborative Servers

Tools

SIP Phone Context

Routing Tests

H.323

SIP

Backup

Restore

GK/NRS Data upgrade

Managing:

Active database

Standby database

47.248.100.141

Numbering Plans > Routes

Search for Routing Entries

Hide

Enter a DnPrefix and Dn Type (use \* for all) and click Search.You may narrow the search by specifying a particular domain.

DN Prefix: \*

DN Type: All DN Types

Limit results to Domain: nortel-dplab.com

udp

cdp

Endpoint Name: Convergys01

Results per page: 50

Search

Routing Entries (1)

Default Routes (0)

Add...

Copy...

Move...

Import...

Export...

Routing test...

Delete

Refresh

	DN Prefix	DN Type	Route Cost	SIP URI Phone Context	Context
1	3200	Private level 0 regional (CDP steering code)	1	cdp.udp	nortel-dplab.com / udp / cdp / Convergys01

1 - 1 of 1 Routing Entry(ies)

Page 1 of 1

First| Previous| Next| Last

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

This section describes the steps on how to create:

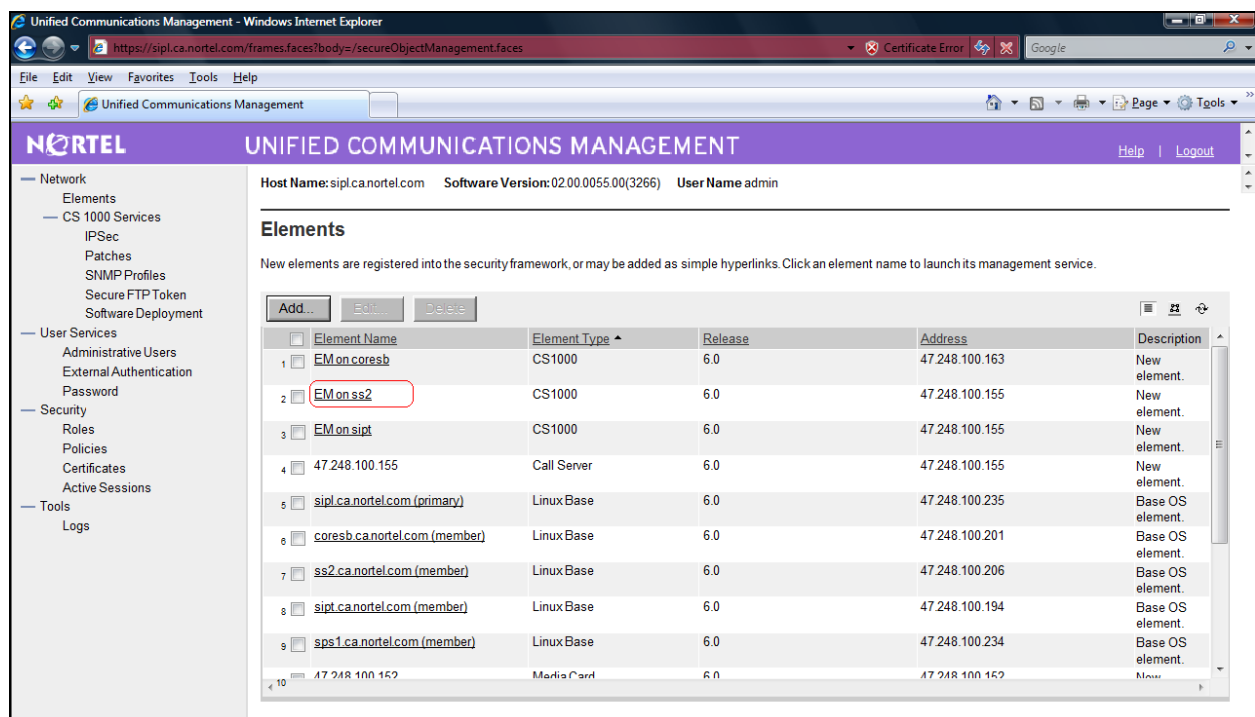
- Route Data Blocks
- Trunks
- ESN Data Block
- Patching

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

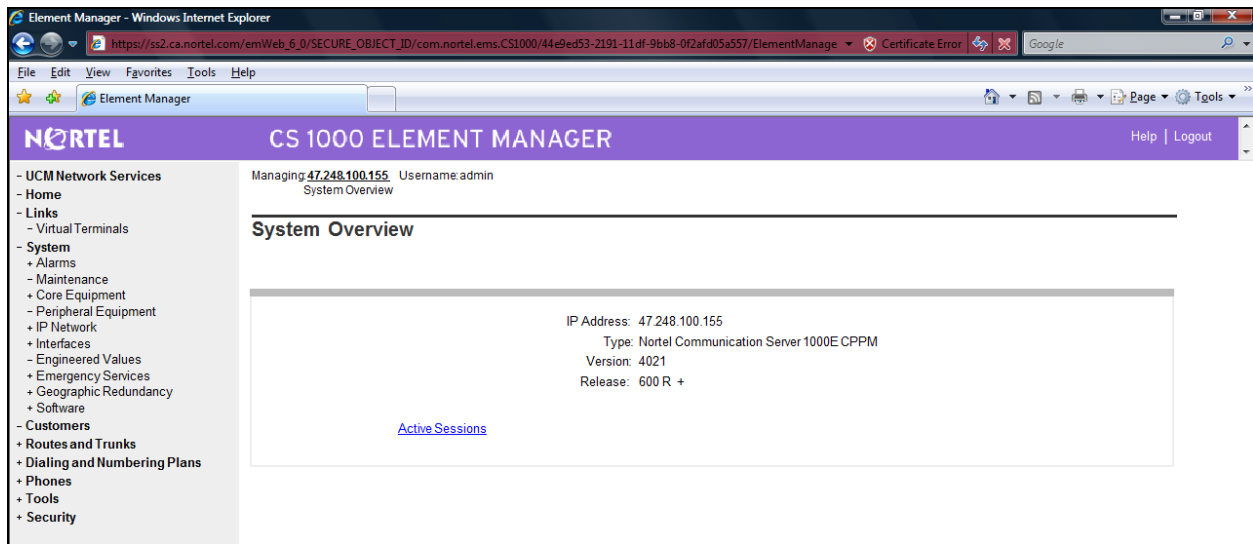
Perform the following steps to register CS 1000 Node ID **555** to the NRS

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.**

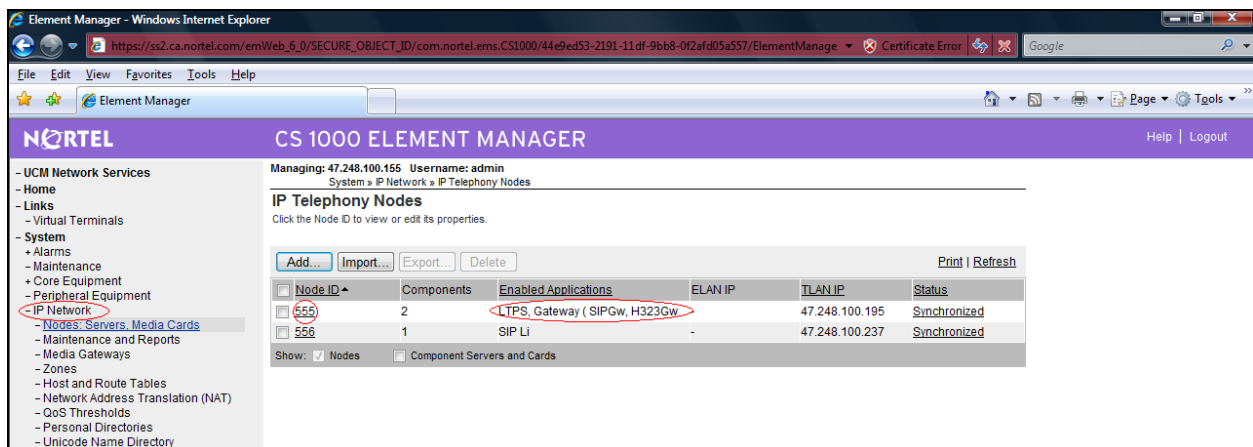
The EM homepage of Call Server appears as shown in Figure 12.



**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.

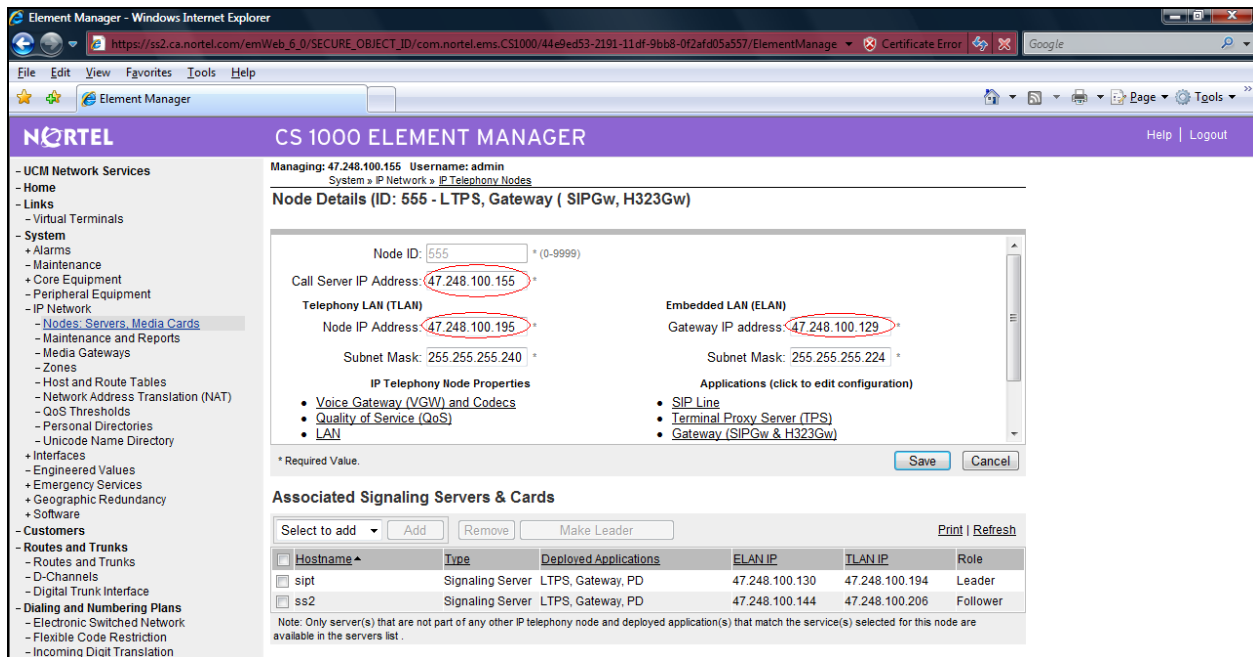
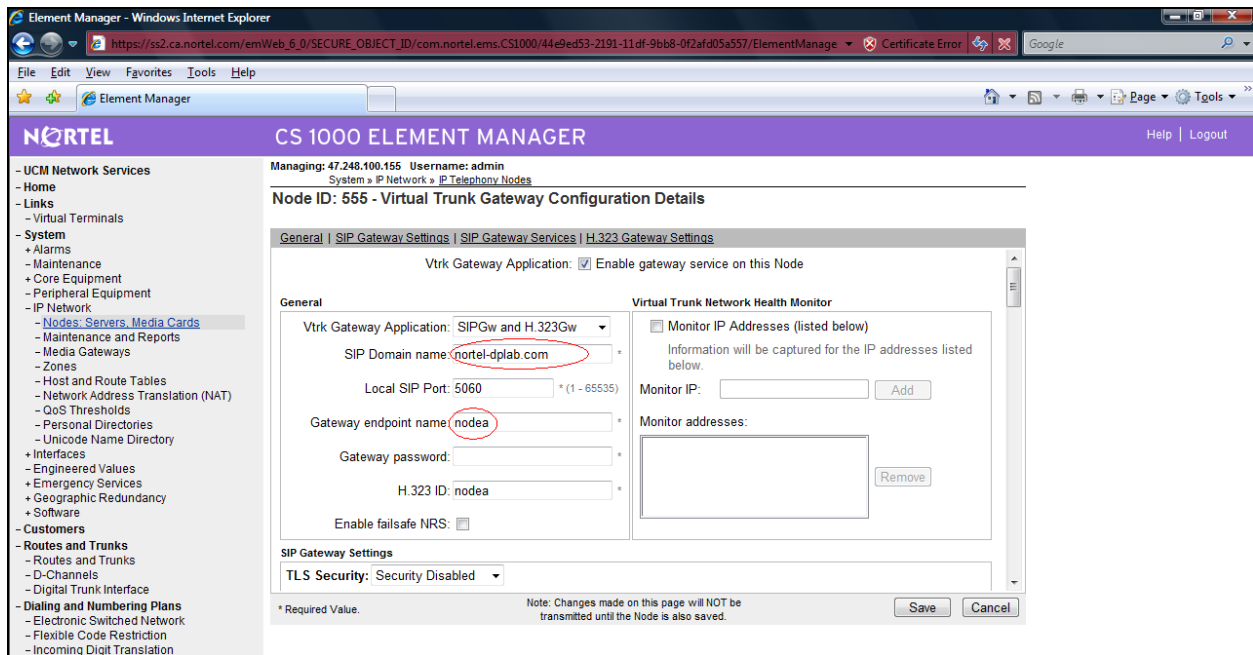


Figure 14: Node ID 555 page on the EM

- 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.
- Notes:** **SIP Domain name:** nortel-dplab.com → 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.

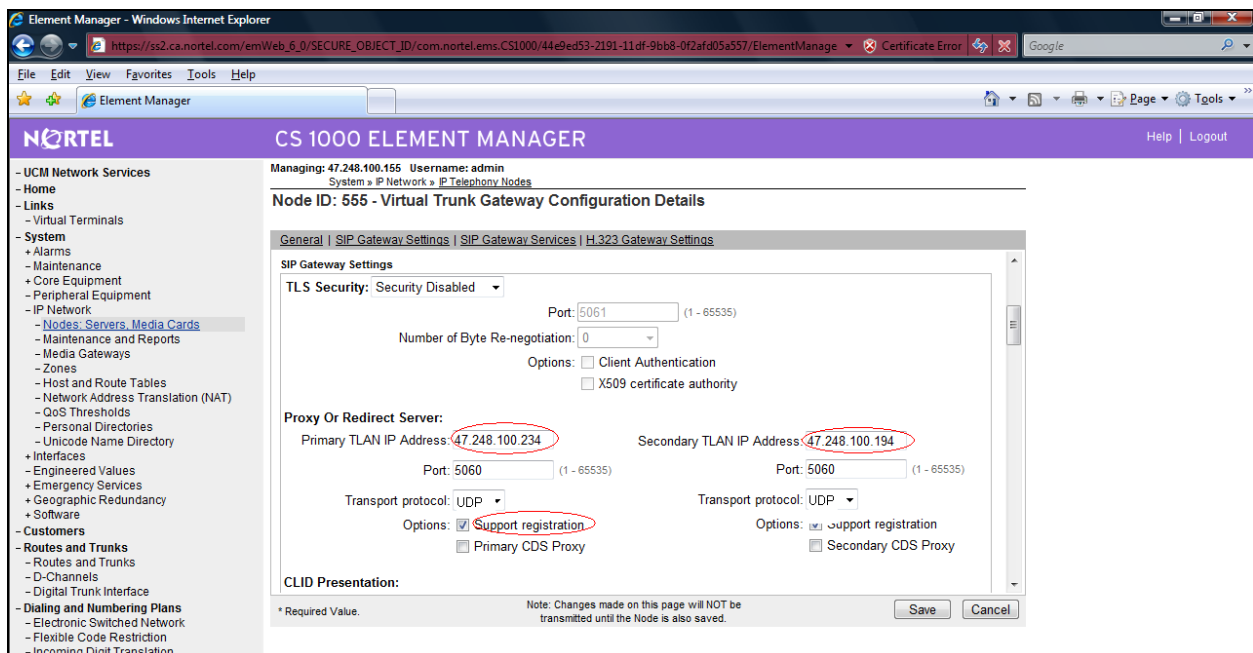


**Figure 15: Virtual Trunk Gateway Configuration Details Page of EM**

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

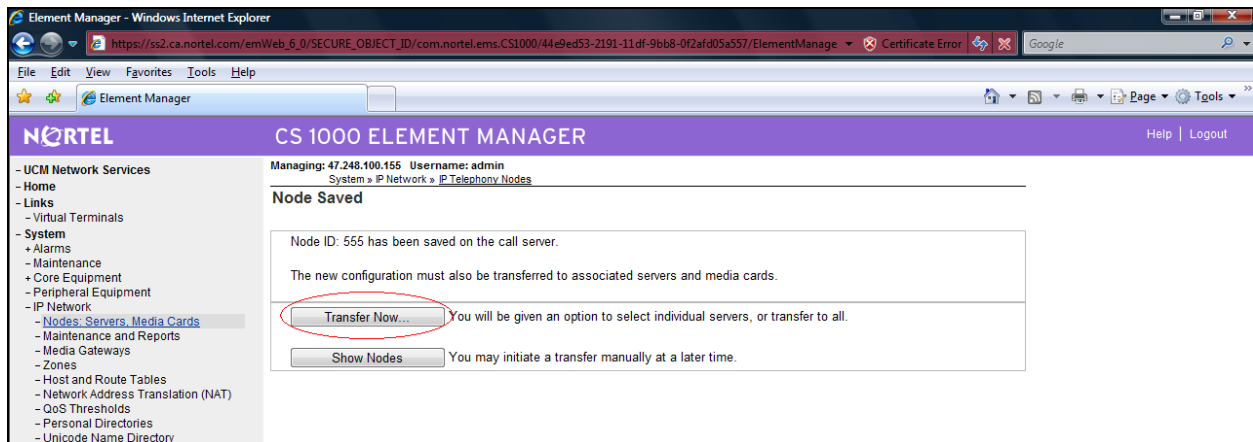
The remaining fields can be left at their default values as shown in Figure 15c. Click on the **Save** button.

**Figure 15c – Virtual Trunk Gateway Configuration Details Page**

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

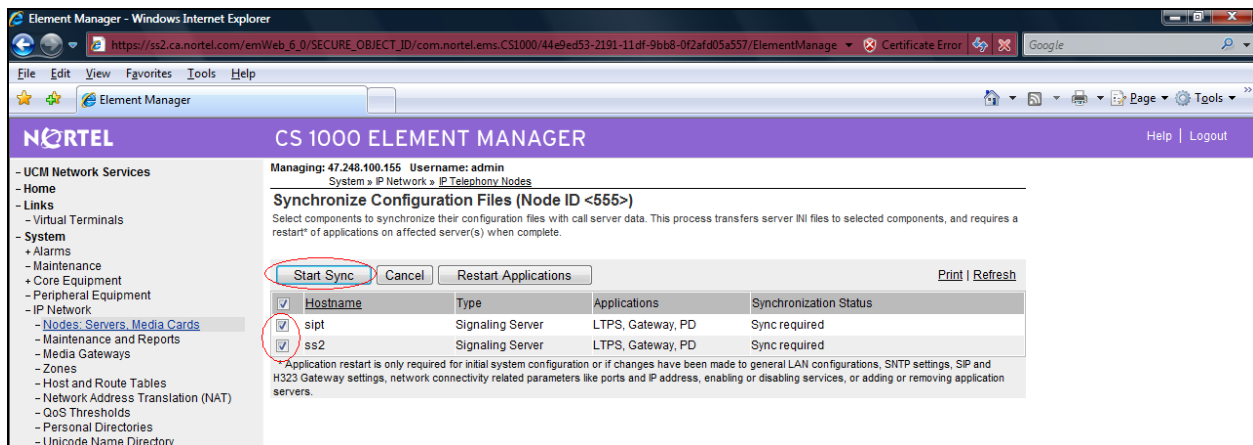
- Click on the **Save** button to save configuration details. When finished, the system will bring back the Node ID page (not shown). Then Click on the **Save** button on the Node ID page and that will take the user to the **Node Saved** page as shown in Figure 16. Click on the **Transfer Now** button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

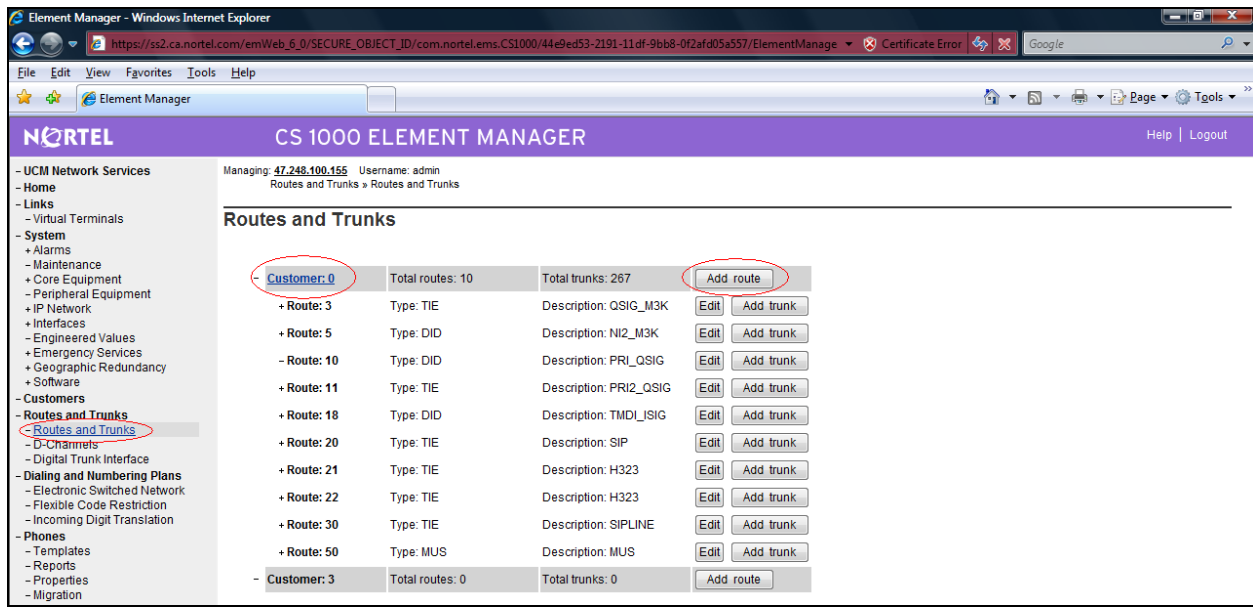


Figure 18: Routes and Trunks webpage

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)**

Element Manager - Windows Internet Explorer

https://ss2.ca.nortel.com/emWeb\_6\_0/SECURE\_OBJECT\_ID/com.nortel.ems.CS1000/44e9ed53-2191-11df-9bb8-0f2af05a557/ElementManager

File Edit View Favorites Tools Help

Element Manager

NORTEL CS 1000 ELEMENT MANAGER Help | Logout

Managing 47.248.100.155 Username: admin  
Routes and Trunks » Routes and Trunks » Customer 0, New Route Configuration

### Customer 0, New Route Configuration

**- Basic Configuration**

Route data block (RDB) (TYPE): RDB

Customer number (CUST): 0

Route number (ROUT): 20

Designator field for trunk (DES): SIPRoute

Trunk type (TKTP): TIE trunk data block (TIE)

Incoming and outgoing trunk (ICOG): Incoming and Outgoing (IAO)

Access code for the trunk route (ACOD): 8900

Trunk type M911P (M911P): ☐

The route is for a virtual trunk route (VTRK): ☒

- Zone for codec selection and bandwidth management (ZONE): 255 (0 - 255)

- Node ID of signaling server of this route (NODE): 555 (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): ☐

**+ Basic Route Options**

**+ Network Options**

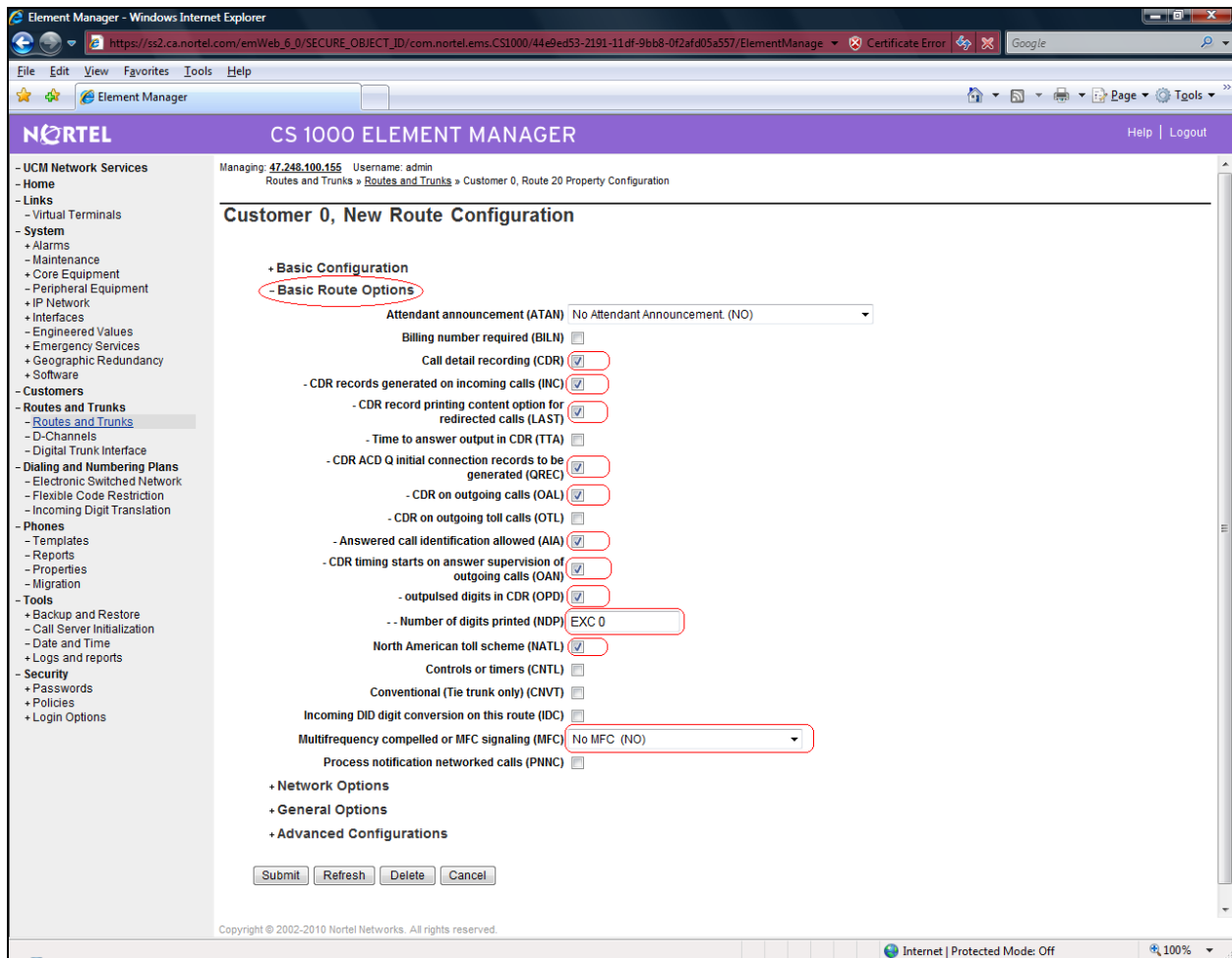
**+ General Options**

**+ Advanced Configurations**

Save Cancel

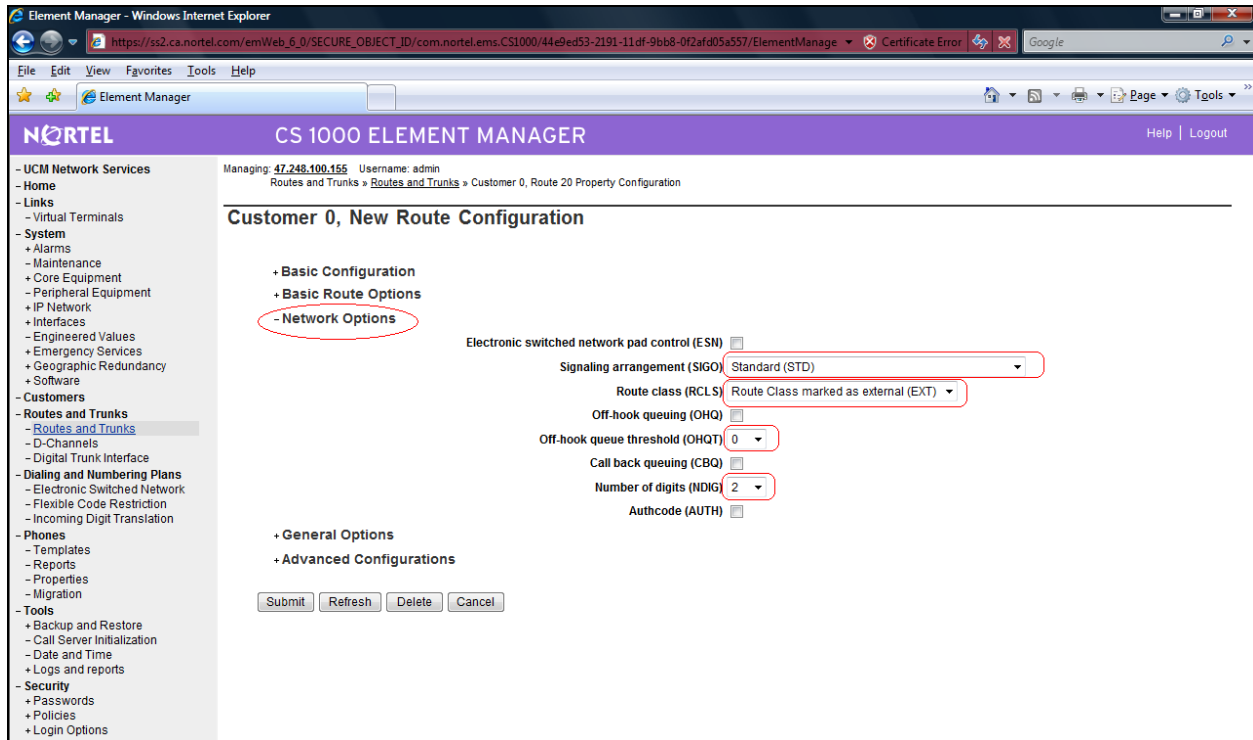
**Figure 19: Basic Configuration of RDB**

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.

Element Manager - Windows Internet Explorer

https://ss2.ca.nortel.com/emWeb\_6\_0/SECURE\_OBJECT\_ID/com.nortel.ems.CS1000/44e9ed53-2191-11df-9bb8-0f2af05a557/ElementManager

File Edit View Favorites Tools Help

Element Manager

NORTEL CS 1000 ELEMENT MANAGER Help Logout

Managing: 47.246.100.155 Username: admin  
Routes and Trunks > Routes and Trunks > Customer 0, Route 20 Property Configuration

### Customer 0, New Route Configuration

- + Basic Configuration
- + Basic Route Options
- + 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) Range: 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) ☐

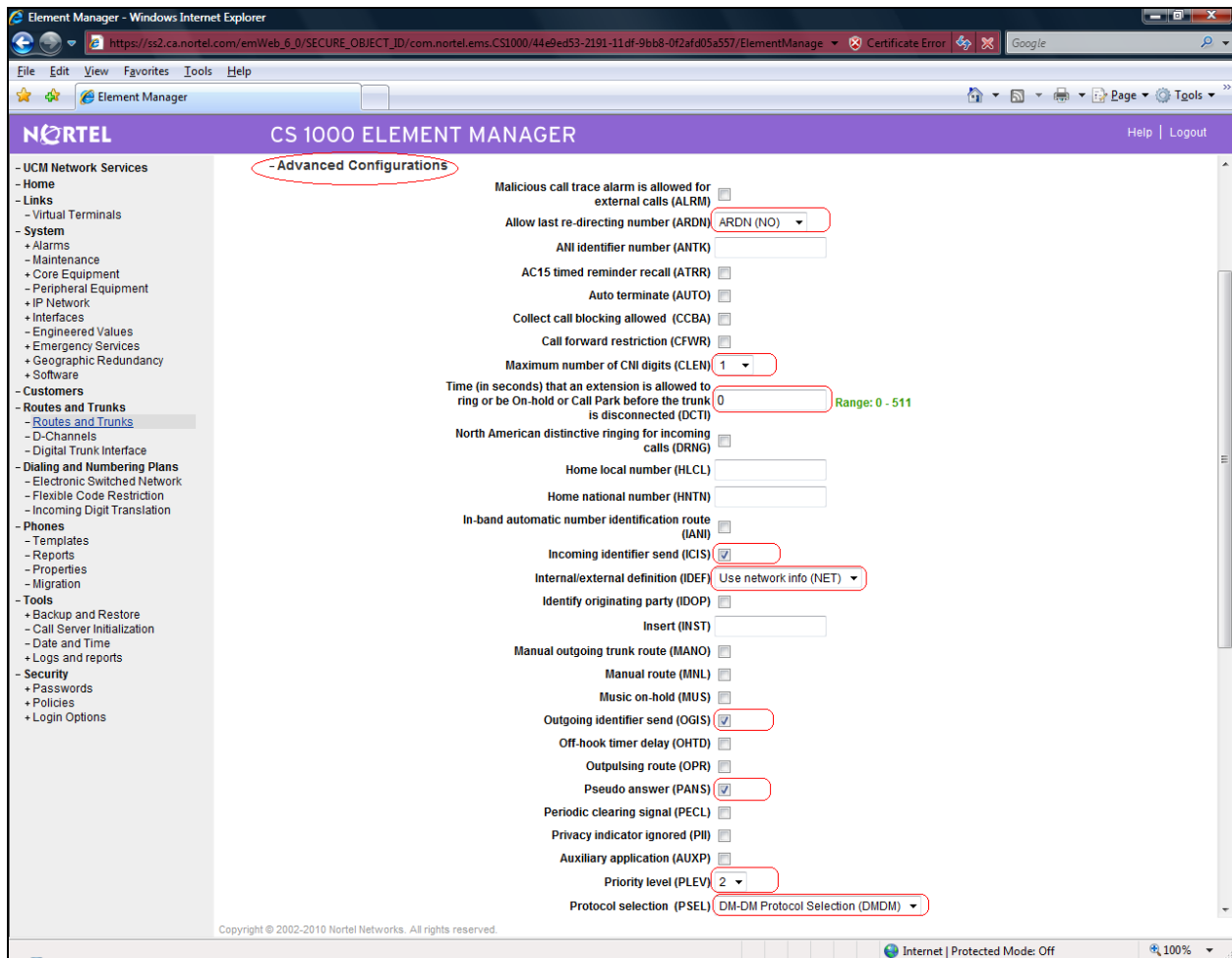
Include DNIS number in CDR records (DCDR) ☐

+ Advanced Configurations

Submit Refresh Delete Cancel

**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**

The screenshot shows the 'CS 1000 ELEMENT MANAGER' web interface. The left sidebar contains a navigation tree with the following items:

- 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 (selected)
      - D-Channels
      - Digital Trunk Interface
    - Dialing and Numbering Plans
      - Electronic Switched Network
      - Flexible Code Restriction
      - Incoming Digit Translation
    - Phones
      - Templates
      - Reports
      - Properties
      - Migration
    - Tools
      - Backup and Restore
      - Call Server Initialization
      - Date and Time
      - Logs and reports
    - Security
      - Passwords
      - Policies
      - Login Options

The main configuration area on the right includes the following settings:

- Music on-hold (MUS) ☐
- Outgoing identifier send (OGIS) ☒
- Off-hook timer delay (OHTD) ☐
- Outpulsing route (OPR) ☐
- Pseudo answer (PANS) ☒
- Periodic clearing signal (PECL) ☐
- Privacy indicator ignored (PII) ☐
- Auxiliary application (AUXP) ☐
- Priority level (PLEV)
- Protocol selection (PSEL) DM-DM Protocol Selection (DMDM)
- Preference trunk usage threshold (PTUT)  Range: 0 - 510
- Port type at far end (PTYP) Analog TIE trunks (ATT)
- Route traffic information in ACD Reports (RACD) ☐
- Radio paging route (RPA) ☐
- Route number (RTN)  Range: 0 - 511
- Satellite used for trunk route (SAT) ☐
- Scheduled access restriction group (SGRP)  Range: 0 - 999
- Special service list number (SSL)
- Standard signaling type (STYP) Standard Data (SDAT)
- CPPI/CPPO flag for incoming non-ISDN trunk call tandemed to this trunk route (TCPP) ☐
- Tone detector required (TDET) ☐
- Trunk identity (TIDY) 8600 20
- Tromboning (TRMB) ☒
- Recall signal (may not) may be received and transmitted on this route (TRRL) ☐
- Tone table number (TTBL)
- Answer an attendant extended call over VNS immediately on the incoming bearer trunk (VRAT) ☐

At the bottom of the configuration area are four buttons: Submit, Refresh, Delete, and Cancel.

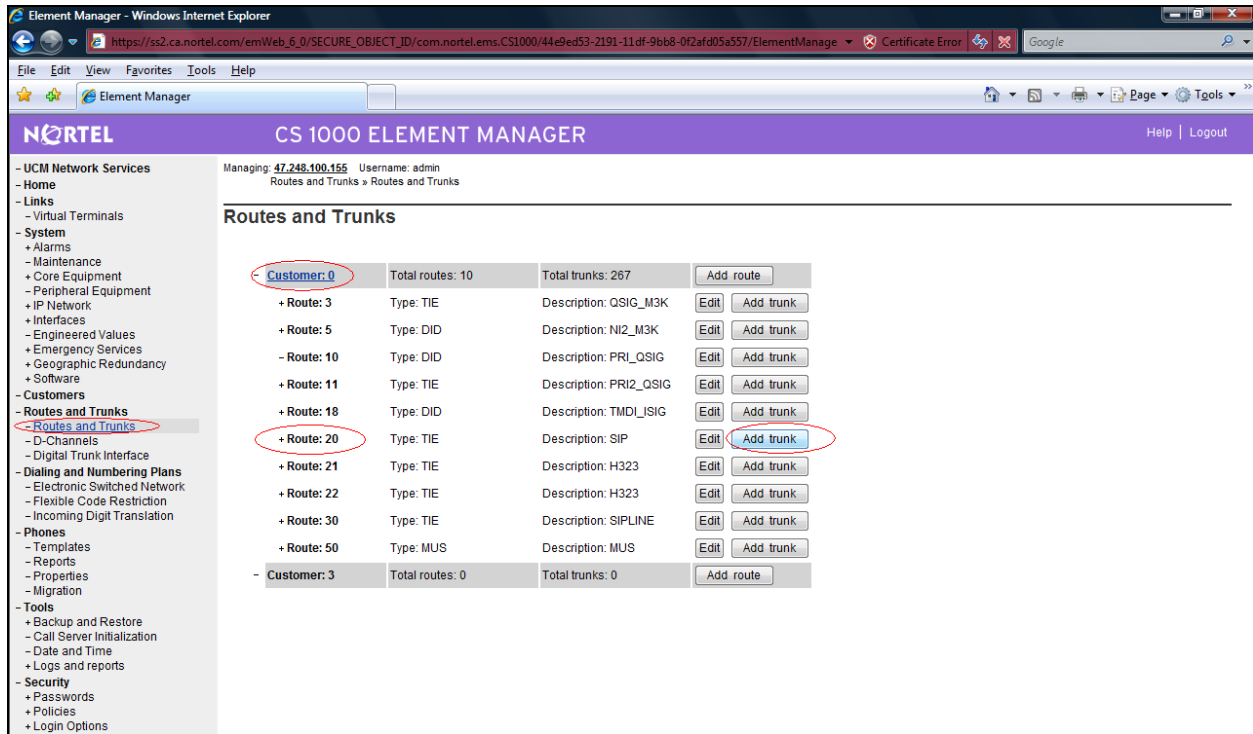
**Figure 23b: Advanced Configurations Section of RDB (Continued)**

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)**.

Managing: 47.246.100.155 Username: admin  
Routes and Trunks » Routes and Trunks » Customer 0, Route 20, New Trunk Configuration

### Customer 0, Route 20, New Trunk Configuration

**- Basic Configuration**

Input Description	Input Value
Multiple trunk input number (MTINPUT)	32
Trunk data block (TYPE)	IP Trunk (IPT)
Terminal Number (TN)	100 0 0 0
Designator field for trunk (DES)	SIP Trunk
Extended Trunk (XTRK)	VTRK
Route number, Member number (RTMB)	20 1
Level 3 Signaling (SIGL)	
Card Density (CDEN)	
Start arrangement Incoming (STR)	Immediate (IMM)
Start arrangement Outgoing (STOR)	Immediate (IMM)
Trunk Group Access Restriction (TGAR)	0
Channel ID for this trunk (CHID)	1
Increase or decrease the member numbers (INC)	Increase channel and member number (YES)
Class of Service (CLS)	Edit

**+ Advanced Trunk Configurations**

Save Cancel

This CHID must be unique in the system

**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.

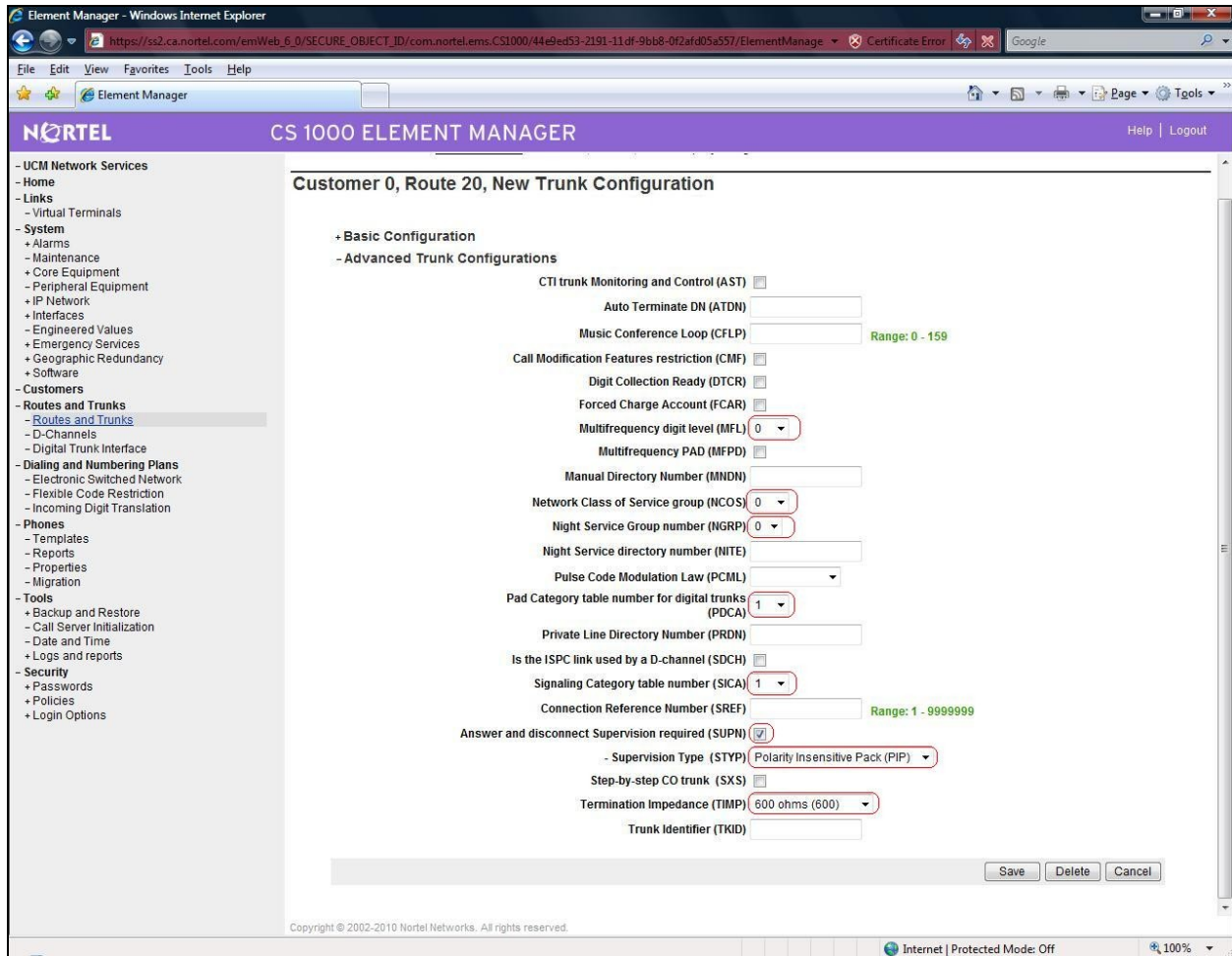
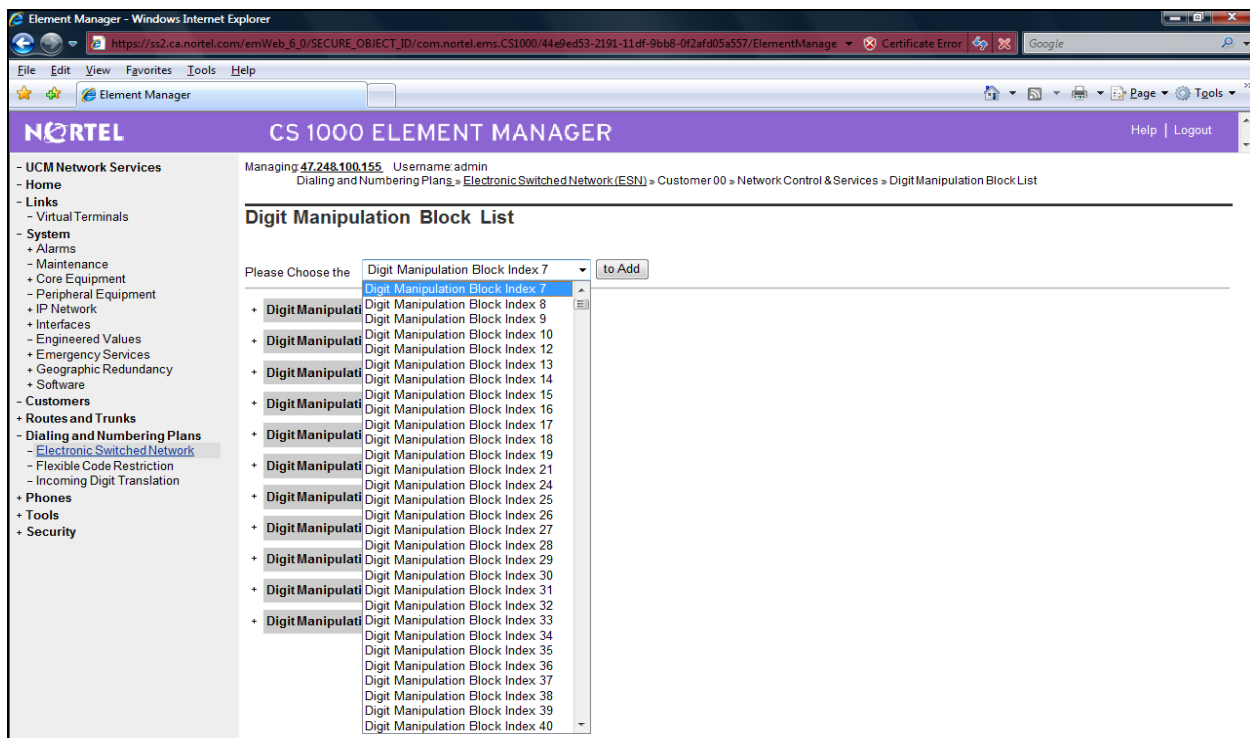


Figure 26: Advanced Trunk Configurations

#### 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.

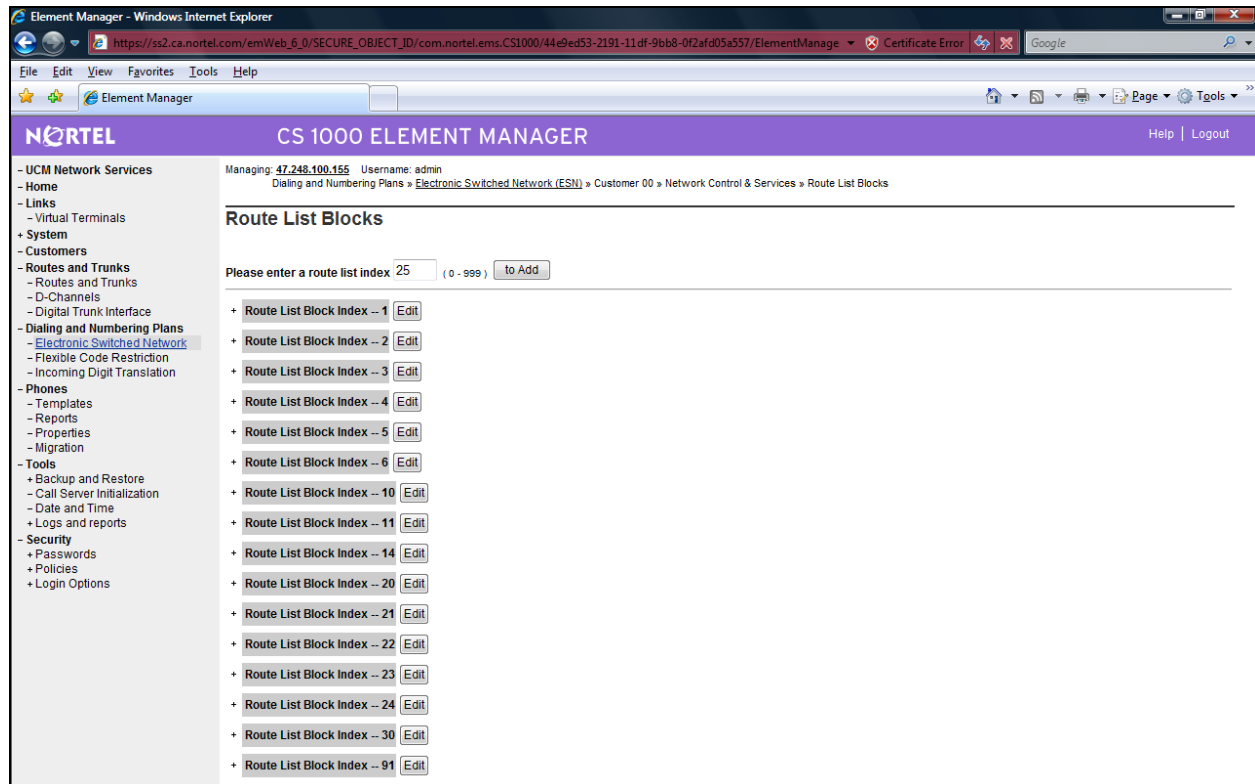


Figure 29: Route List Blocks page

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.

The screenshot shows the 'Route List Block' configuration page in the Nortel CS 1000 Element Manager. The page is divided into a left sidebar with navigation links and a main content area with configuration fields. The configuration fields are organized into a table with two columns: 'Input Description' and 'Input Value'. The fields include:

Input Description	Input Value
Route List Index (RLI):	25
Entry Number for the Route List (ENTR):	0 (0 - 63)
Local Termination entry (LTER):	<input type="checkbox"/>
Route Number (ROUT):	20
Skip Conventional Signaling (SCNV):	<input type="checkbox"/>
Use Tone Detector (TDET):	<input type="checkbox"/>
Time of Day Schedule (TOD):	0
Entry is a VNS Route (VNS):	<input type="checkbox"/>
Conversion to LDN (CNV):	<input type="checkbox"/>
Expensive Route (EXP):	<input type="checkbox"/>
Facility Restriction Level (FRL):	0 (0 - 7)
Digit Manipulation Index (DMI):	7
ISL D-Channel Down Digit Manipulation Index (ISDM):	0 (0 - 999)
Free Calling Area Screening Index (FCI):	0
Free Special Number Screening Index (FSNI):	0
Business Network Extension Route (BNE):	<input type="checkbox"/>
Strategy on Congestion (SBOC):	No Reroute (NRR)
QSIG Alternate Routing Causes (COPT):	QSIG Alternate Routing Cause 1
ISDN Drop Back Busy (IDBB):	Drop Back Disabled (DBD)
ISDN Off-Hook Queuing Option (IOHQ):	<input type="checkbox"/>
Off-Hook Queuing Allowed (OHQ):	<input type="checkbox"/>
Call Back Queuing Allowed (CBQ):	<input type="checkbox"/>
Number of Alternate Routing Attempts (NALT):	5 (1 - 10)
Initial Set (ISET):	0 (0 - 64)
Set Minimum Facility Restriction Level (MFRL):	
Overlap Length (OVLL):	0 (0 - 24)

At the bottom of the page, there are 'Submit' and 'Cancel' buttons, and a copyright notice: 'Copyright © 2002-2010 Nortel Networks. All rights reserved.'

Figure 30: Detail of Route List Block page

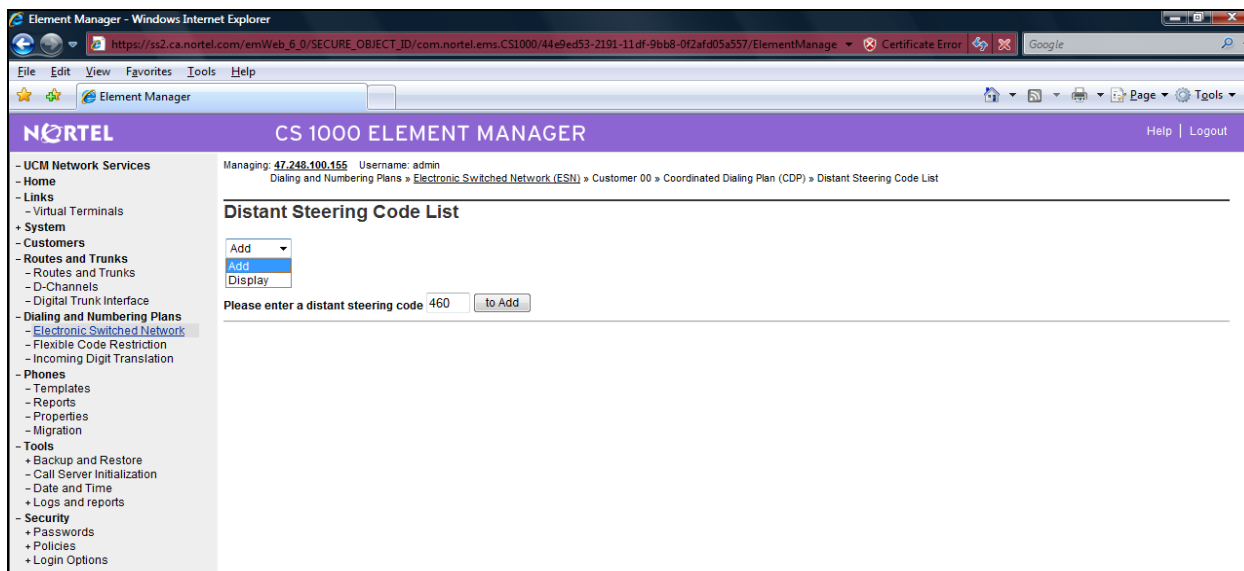
#### 4.3.6. Creating Coordinated Dialing Plan (CDP) on the EM

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**

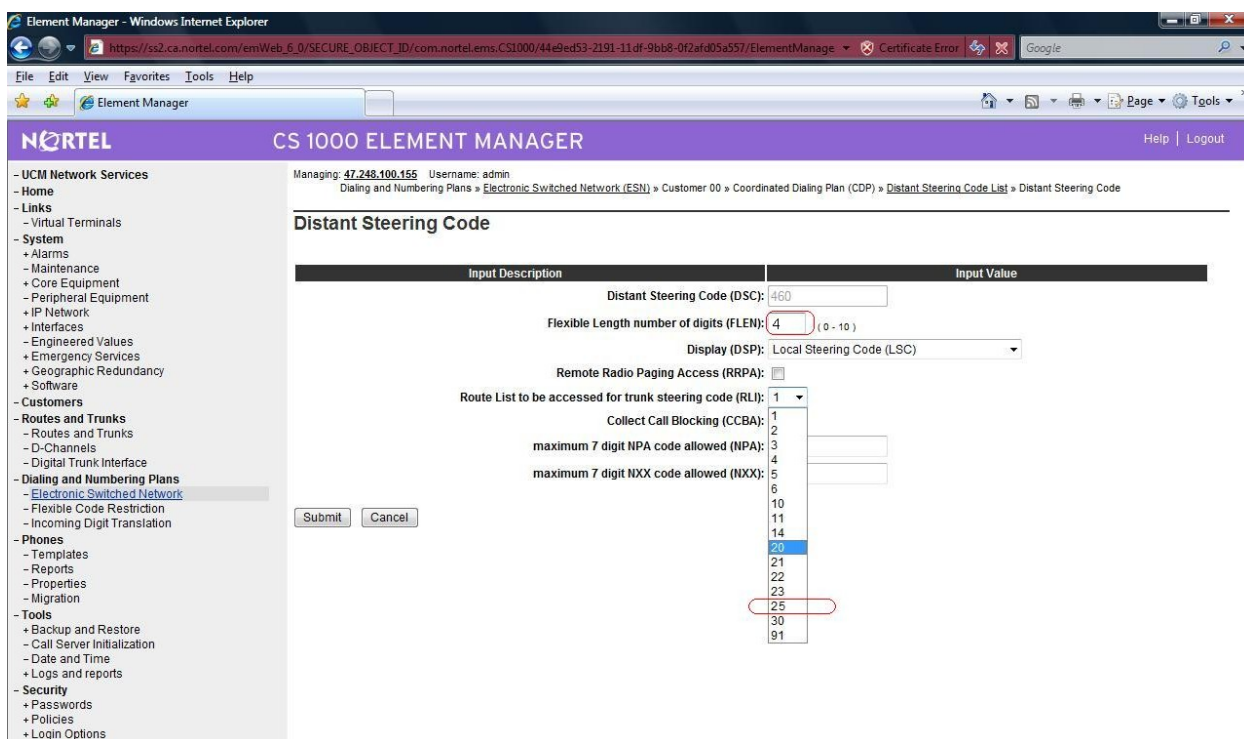
The **Distant Steering Code** details page will appear as shown in Figure 32.

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.



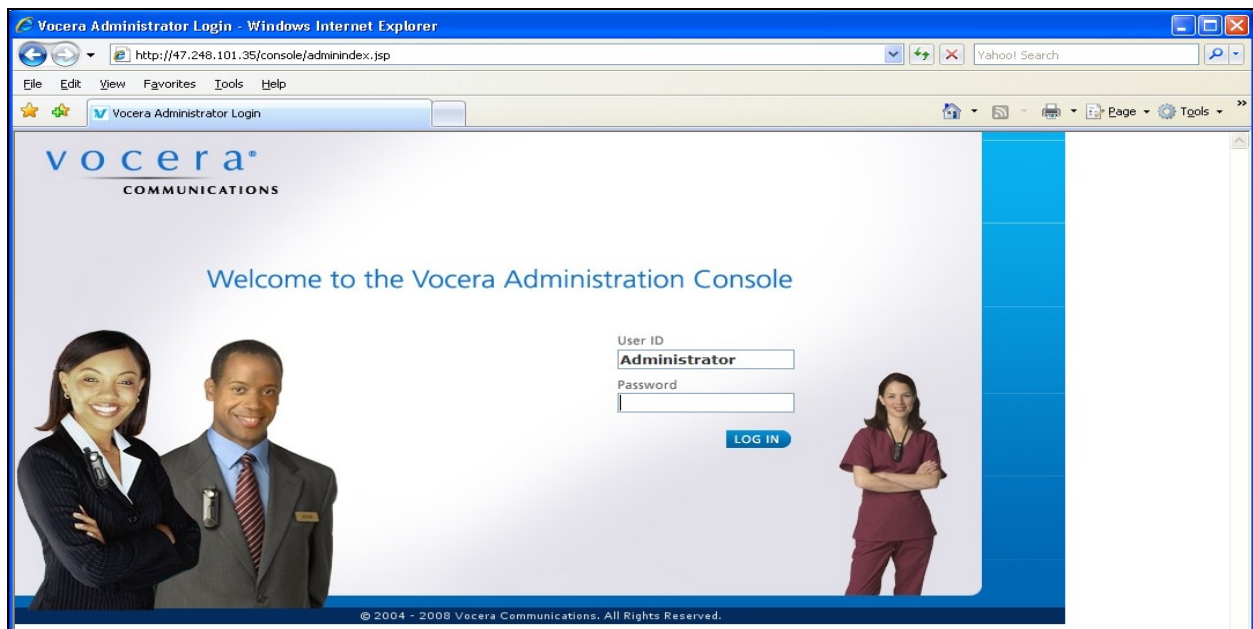
**Figure 32: Detail of Distant Steering Code page**

## 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

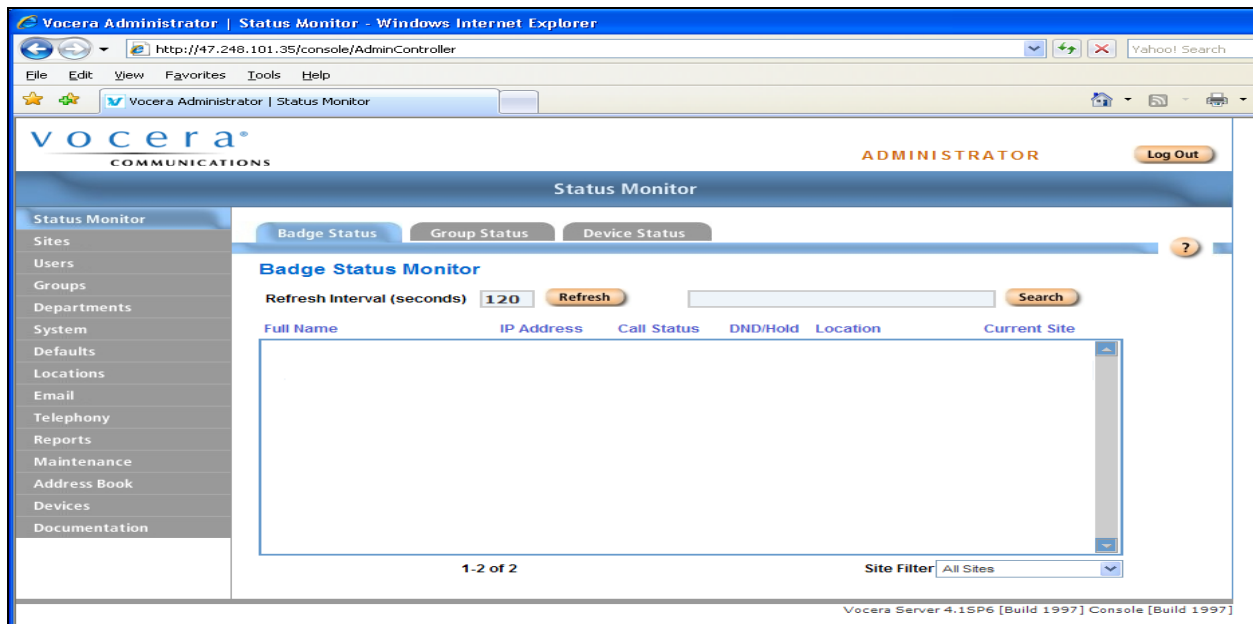
Open the Vocera Communication Systems web page by addressing the IP address of the Vocera Server in the Microsoft Internet browser, <http://47.248.101.35/>. The Welcome to Vocera page will appear (Not shown), then click on the Vocera Administration Console link to get to the console web page as shown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

Vocera Administrator | Telephony - Windows Internet Explorer

http://47.248.101.35/console/SiteController

Vocera Administrator | Telephony

Vocera<sup>®</sup> COMMUNICATIONS

ADMINISTRATOR Log Out

Telephony

Basic Info Access Codes Toll Info DID Info PIN Dynamic Extensions Sharing

Select Site Global

☒ Enable Telephony Integration

Vocera Hunt Group Numbers

Guest Access 3200

Direct Access 3201

Number of Lines\* 6

Integration Type

☐ Analog

☐ Digital

☒ IP

Note: Saving any changes to digital parameters will cause the telephony server to restart.

IP Settings

Signaling Protocol SIP Version 2.0

SIP Settings

Call Signaling Address nortel-dplab.com

Calling Party Number 3200

**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.

Vocera Administrator | Telephony - Windows Internet Explorer

http://47.248.101.35/console/SiteController#

Vocera Administrator | Telephony

**vocera**  
COMMUNICATIONS

ADMINISTRATOR [Log Out](#)

Telephony

Basic Info | **Access Codes** | Toll Info | DID Info | PIN | Dynamic Extensions | Sharing

Select Site: Global

Local Area Code\*  ☐ Omit Area Code when Dialing Locally

Default Local Access Code  Default Long-Distance Access Code

Company Voicemail Access Code

**Access Code Exceptions**

By default, numbers in the local area code use the Default Local Access Code and all others use the Default Long-Distance Access Code. Enter exceptions in the table below:

Area Code	Range of Numbers	Access Code

[Add](#) [Edit](#) [Delete](#)

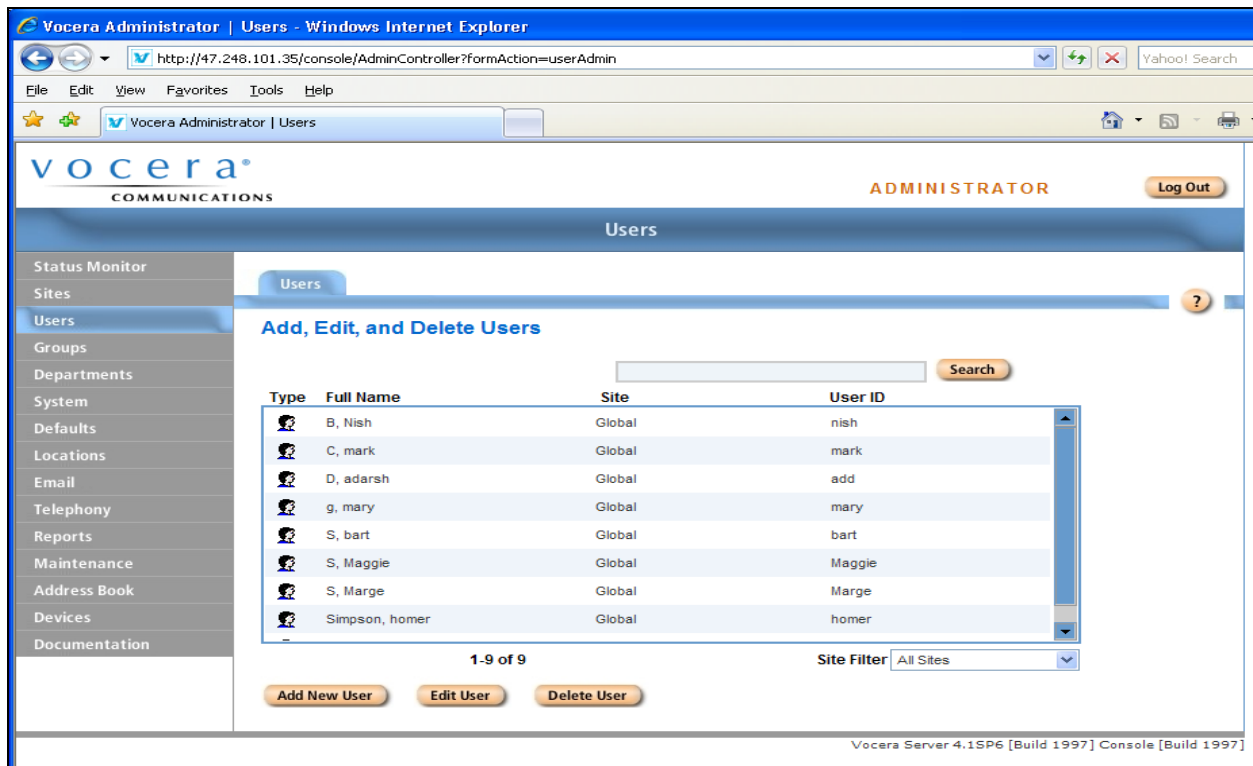
[Save Changes](#) [Reset](#)

Vocera Server 4.1SP6 [Build 1997] Console [Build 1997]

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 shown in 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.

**Add/Edit User -- Webpage Dialog**

http://47.248.101.35/console/admin/adduserdialogframe.jsp

### Add New User

Info Phone Speech Rec Groups Depts Inner Circle ?

First Name \* Last Name \*

Bart Simpson

User ID \* Employee ID

BartS

Password Re-enter Password

Email Address Site

Global Select C

Cost Center Badge ID

0009ef052f8b

☐ Temporary User

Expiration Date (mm/dd/yyyy)

**Note:** Temporary users are removed from the system by the first message sweep after midnight on the expiration date.

Save Save & Continue Cancel

http://47.248.101.35/console/admin/adduserdialogframe.jsp Internet

**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.

The screenshot shows a web browser window titled "Add/Edit User -- Webpage Dialog" with the URL "http://47.248.101.35/console/admin/adduserdialogframe.jsp". The main heading is "Add New User". Below the heading are tabs: "Info", "Phone" (selected), "Speech Rec", "Groups", "Depts", and "Inner Circle". A help icon (?) is visible next to the tabs.

The "Phone" tab contains the following fields:

- Desk Phone or Extension:** A text box containing "5546".
- Cell Phone:** An empty text box.
- Home Phone:** A text box containing "6139564004".
- Pager:** An empty text box.
- Vocera Extension:** An empty text box.
- Dynamic Extension:** An empty text box.
- PIN for Long Distance Calls:** An empty text box.

Below these fields is a section titled "Genie Access from Phone" with a checkbox labeled "Enable Access to Genie from Phone". Below the checkbox are two text boxes: "Phone Password (minimum 5 chars.)" and "Re-enter Phone Password". A note below these boxes states: "Note: Phone password not required if caller ID permission is used."

At the bottom of the dialog are three buttons: "Save", "Save & Continue", and "Cancel". The browser's status bar at the bottom shows the URL "http://47.248.101.35/console/admin/adduserdialogframe.jsp" and an "Internet" icon.

**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>



---

**©2011 Avaya Inc. All Rights Reserved.**

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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](mailto:devconnect@avaya.com).