Avaya Solution & Interoperability Test Lab

Application notes for Convergys Intervoice Media Server 4.0 with 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 Convergys Intervoice Media Server 4.0. During the compliance testing, the Convergys Media Server was able to register to the SIP proxy Server of the Communication Server 1000 via SIP trunk. The Convergys Intervoice Media Server was able to receive and re-direct the calls based on the interactive voice response options that are inputted by callers from the Communication Server 1000 Release IP and SIP Phones. The compliance testing was focused on interoperability of between the Avaya Communication Server 1000 and the Convergys Intervoice Media Server.

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 Convergys Intervoice Media Server release 4.0.1.280 (hereafter referred to as Convergys MS) during the compliance testing. The Convergys MS was tested to make sure all features properly functioned and interoperated with CS 1000.

1.1. Interoperability Compliance Testing

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

- SIP Registration of the Convergys MS to the Avaya CS 1000 NRS SIP Proxy Server via SIP trunk
- Basic SIP trunk calls
- DTMF 2833 and INBAND transmission methods.
- Transfer call to SIP and non-SIP telephones, ACD queue, and the emulated PSTN via PRI trunk.
- Telephony features: Call forward to voicemail with Message Waiting Indication (MWI) notification, busy, hold and retrieve.
- Codec negotiations G.711, G.729, and G723.
- Serviceability.

1.2. Support

For technical support on Convergys MS, please contact Convergys technical support at website www.Convergys.com or telephone: +1-800-344-3000 or email marketing@convergys.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 Convergys MS.

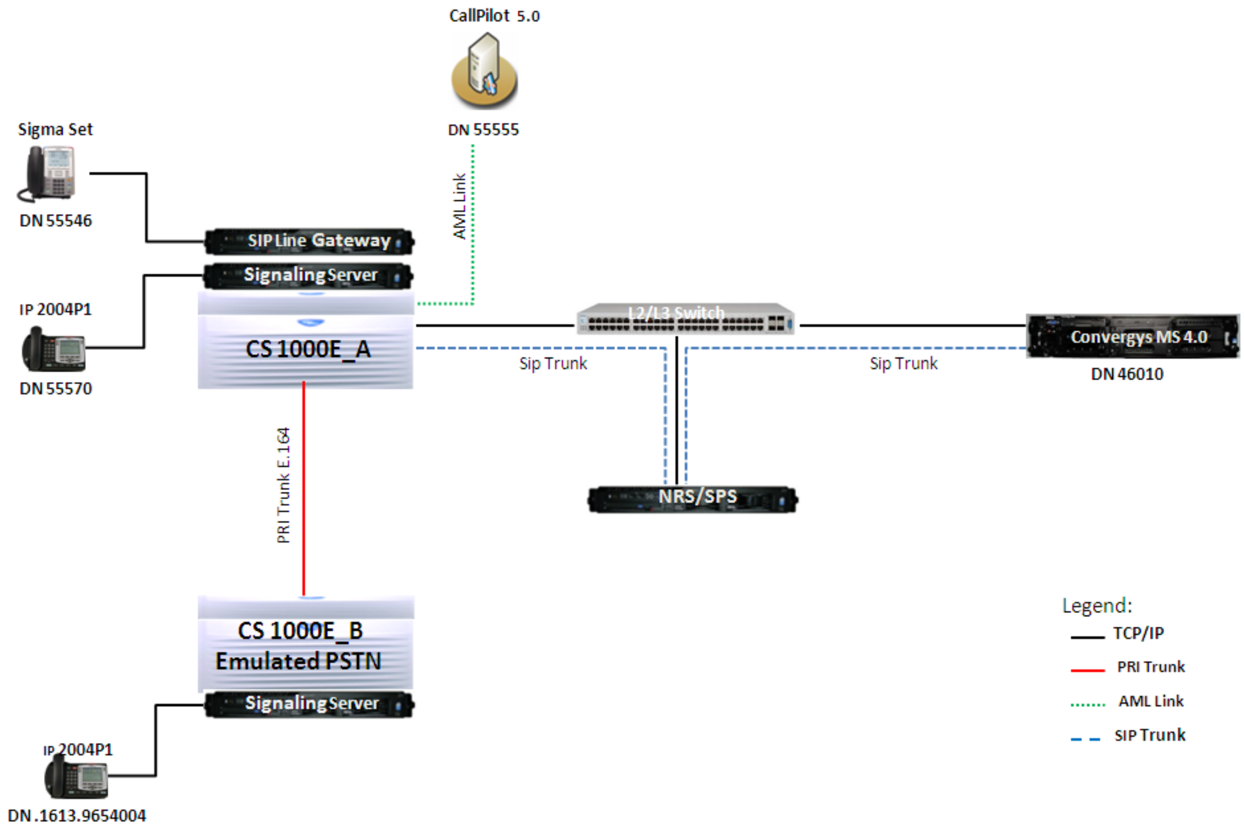

Figure 1: Network Configuration

3. Equipment and Software Validated

System	Software Version
Avaya CS 1000	• Call Server (CPPM): 6.00RJ • Signaling Server (CPPM): 6.00.18 • SIP Line Gateway (HP DL320): 6.00.18
Avaya CallPilot voicemail system	• 5.0
Avaya 11xx SIP client (Sigma) Avaya IP 20042 Avaya IP phones	• 02.02.21.00 • 0602B76 • 2050PC: 3.02.0045
Convergys MS	• 4.0.1.280

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.

Prerequisite

- 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 see [4].

- 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.

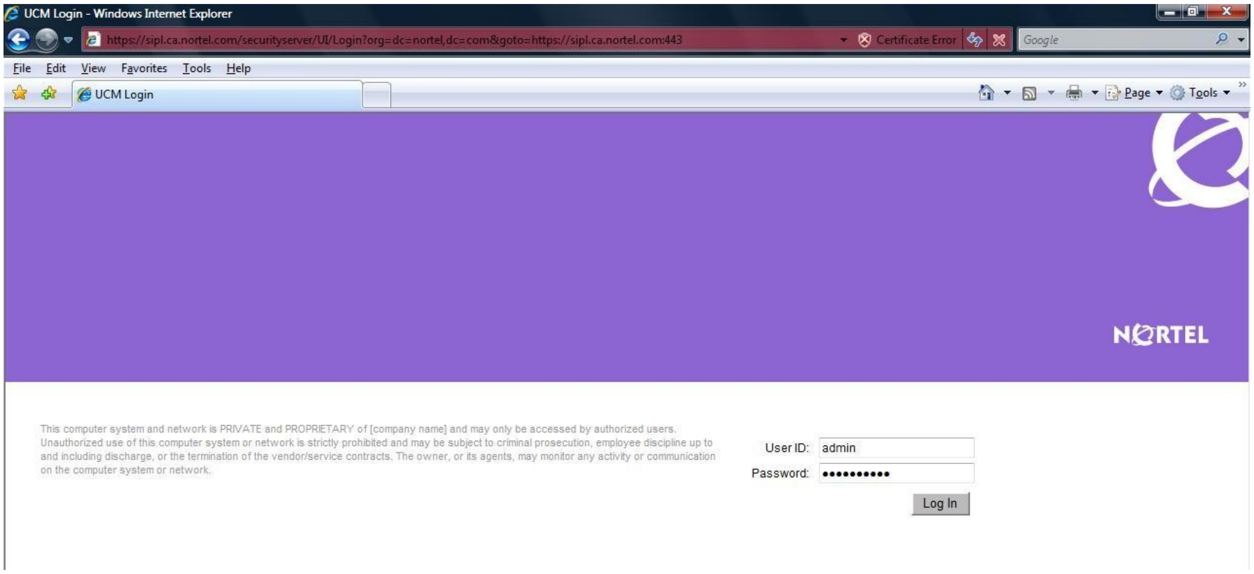

Figure 2: UCM Login web page

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 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 is 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 Convergys MS 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.

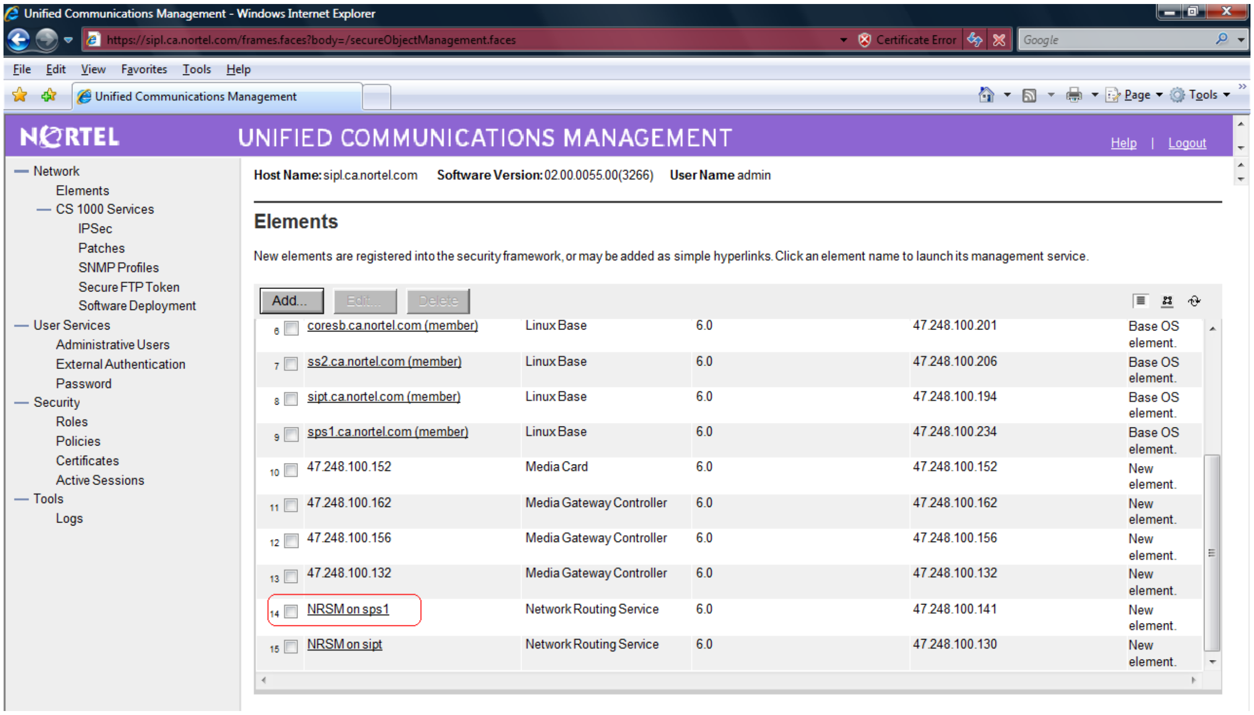

Figure 3: UCM Navigator Homepage

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.

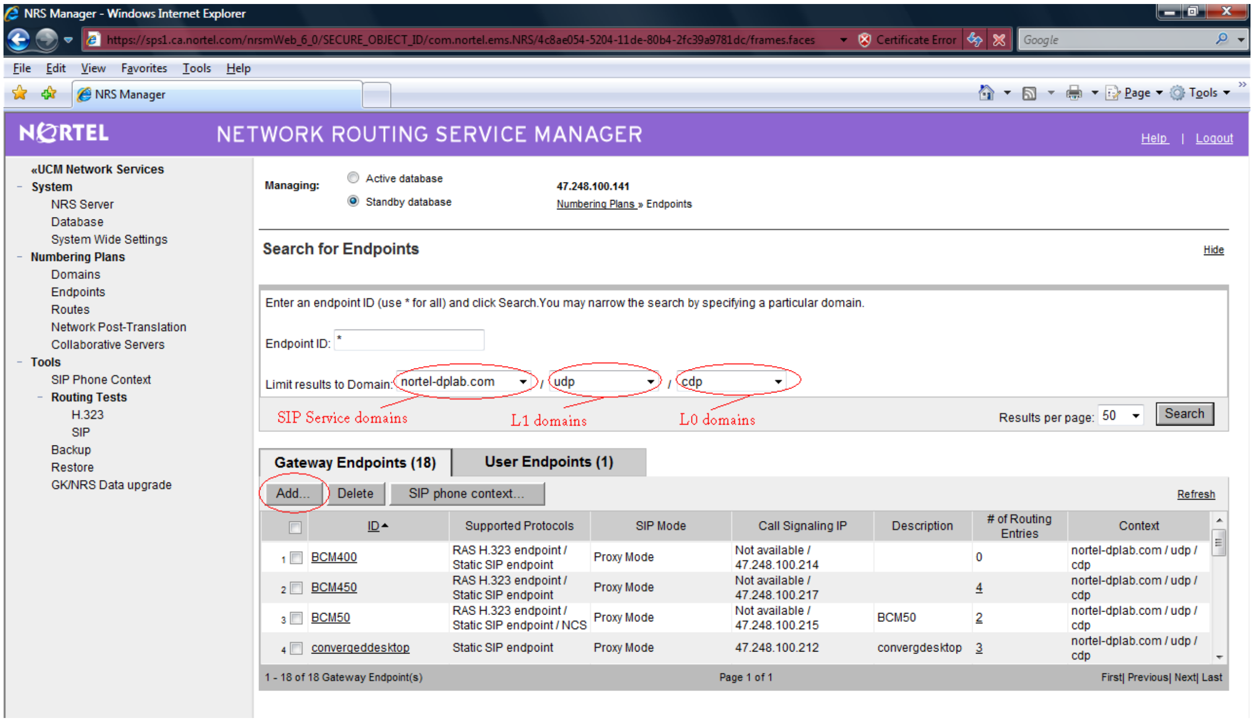

Figure 4: Adding a Gateway Endpoint

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

To configure gateway endpoint, all the highlighted attributes in red circles should be altered as shown in Figure 5. Others are left at default values.

Note: Endpoint name: ***Convergys01*** **→** this name has to be matched with the name of Convergys MS server under test.

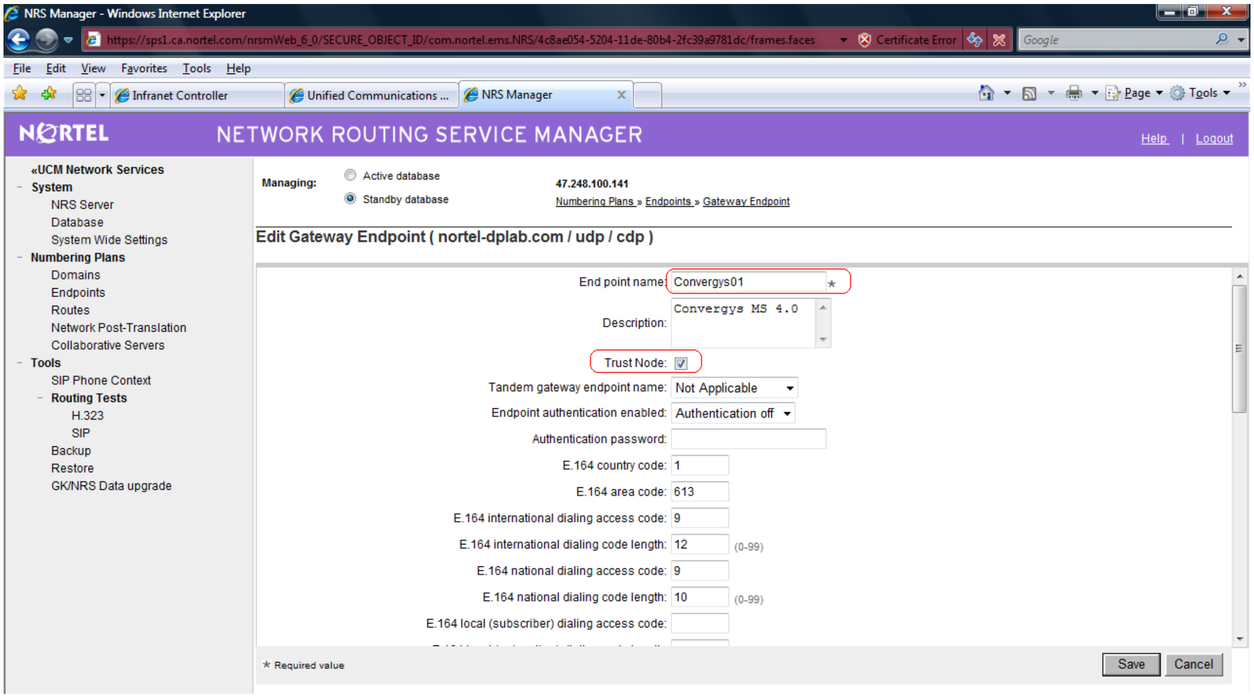

Figure 5: Input Fields of the Convergys Gateway Endpoint

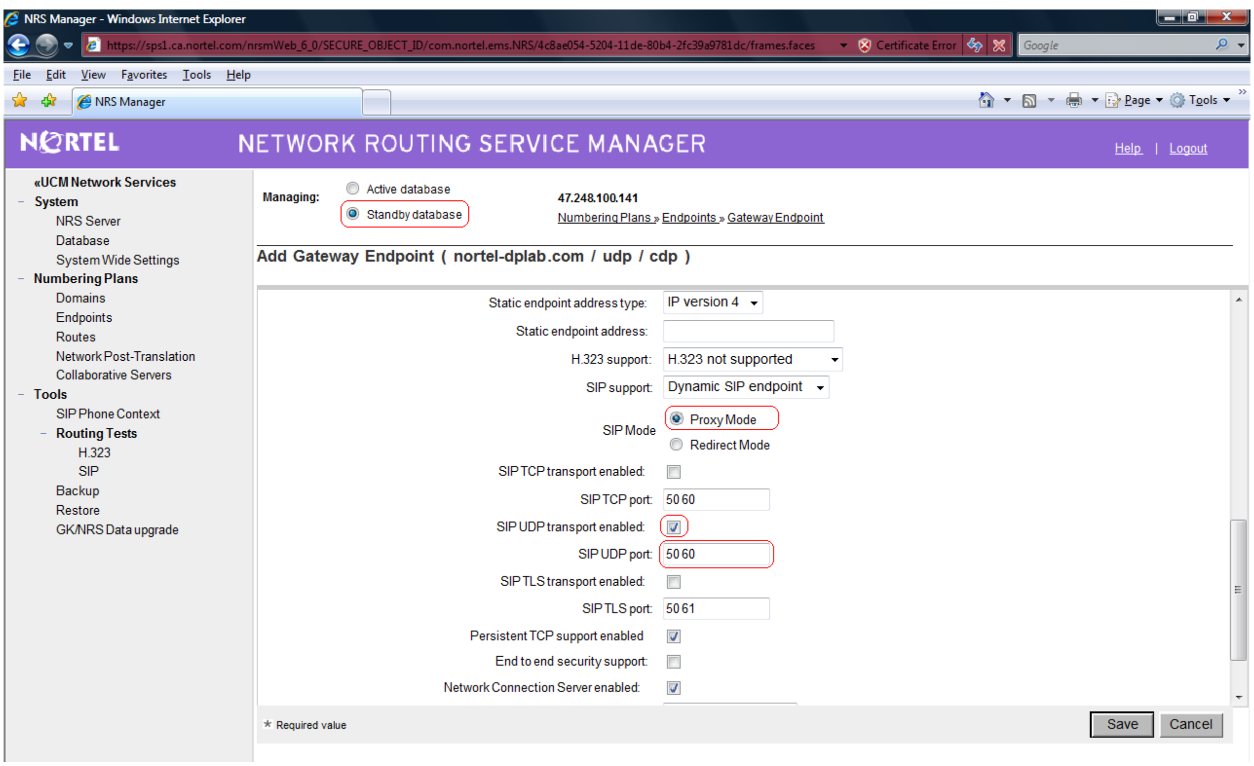

Figure 5b: Input Fields of the Convergys Gateway Endpoint (Continued)

Click on the **Save** button in order to complete the newly created gateway endpoint for Convergys MS. Select the **Database** on the left column under the **System** menu, **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**.

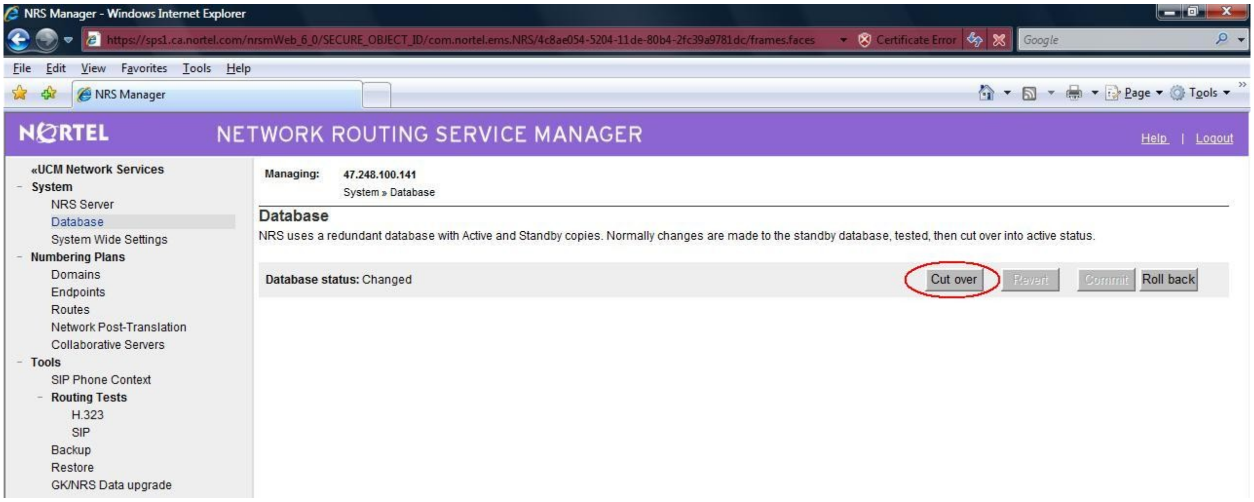

Figure 6: Cut over for Database on NRS

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

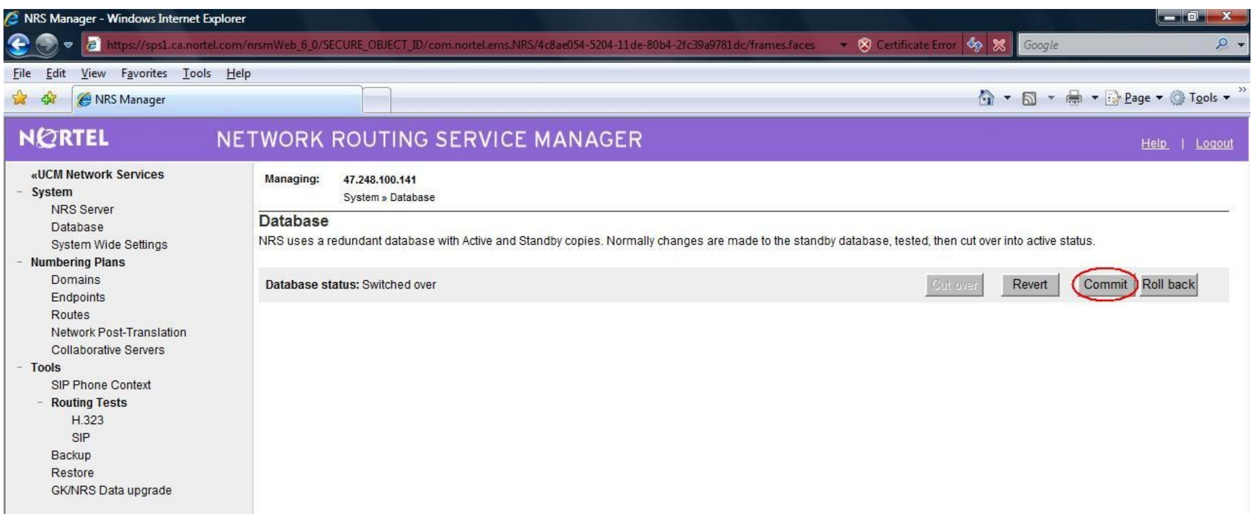

Figure 7: Commit the Database of NRS

4.2.2. Creating a routing entry for Convergys01 gateway endpoint on the NRS

To create a route entry for the **Convergys01** 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 **Convergys01** endpoint name in the list of gateway endpoints.
- Then click on the **Add** button.

<u>**Note**</u>: The **Convergys01** endpoint name only shows up on the list of gateway endpoints when its L1 and L0 domains are selected.

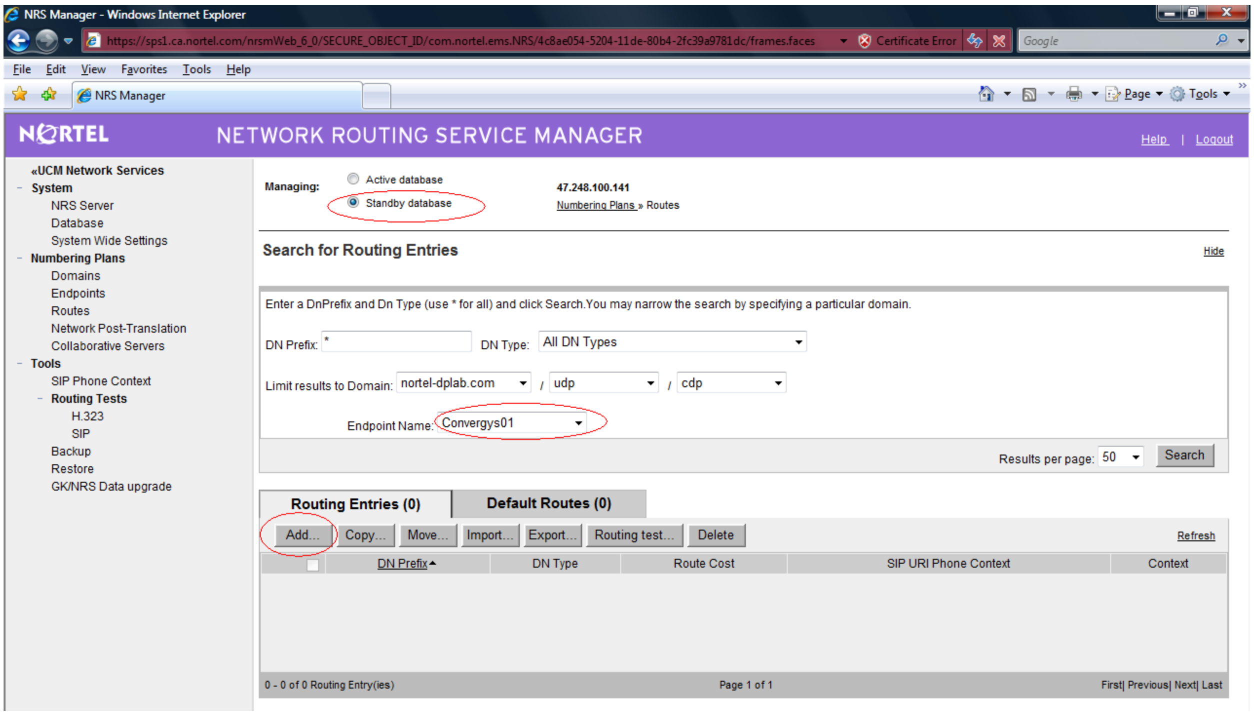

Figure 8: Adding Route Entry for Convergys Endpoint

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

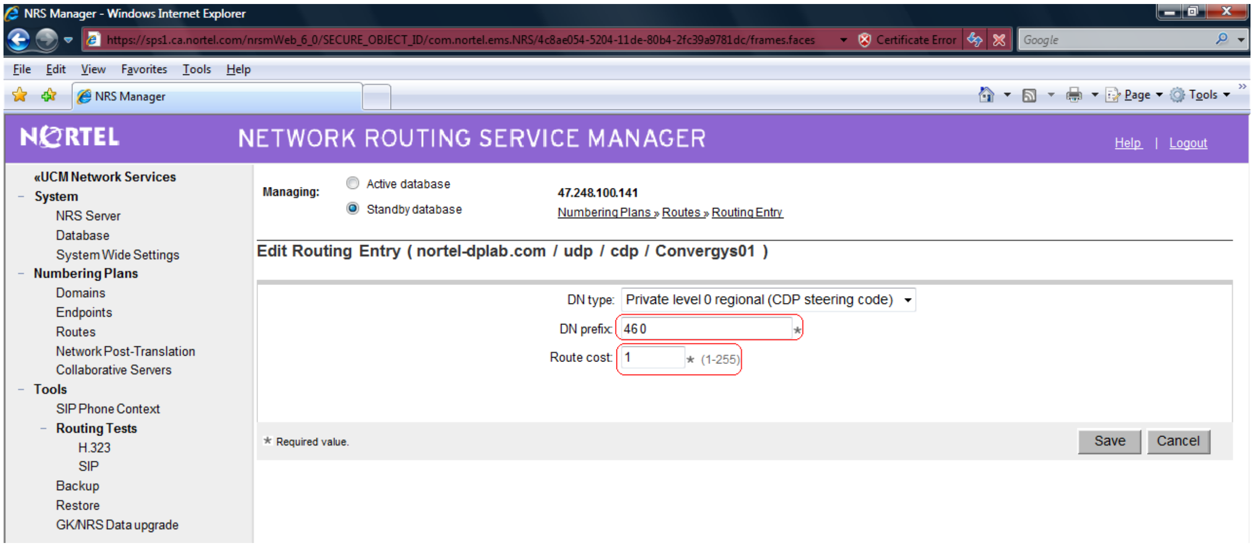

Figure 9: Configure Route Entry for Convergys Endpoint

Similarly in creating the Convergys 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 **460** of the **Convergys01** endpoint.

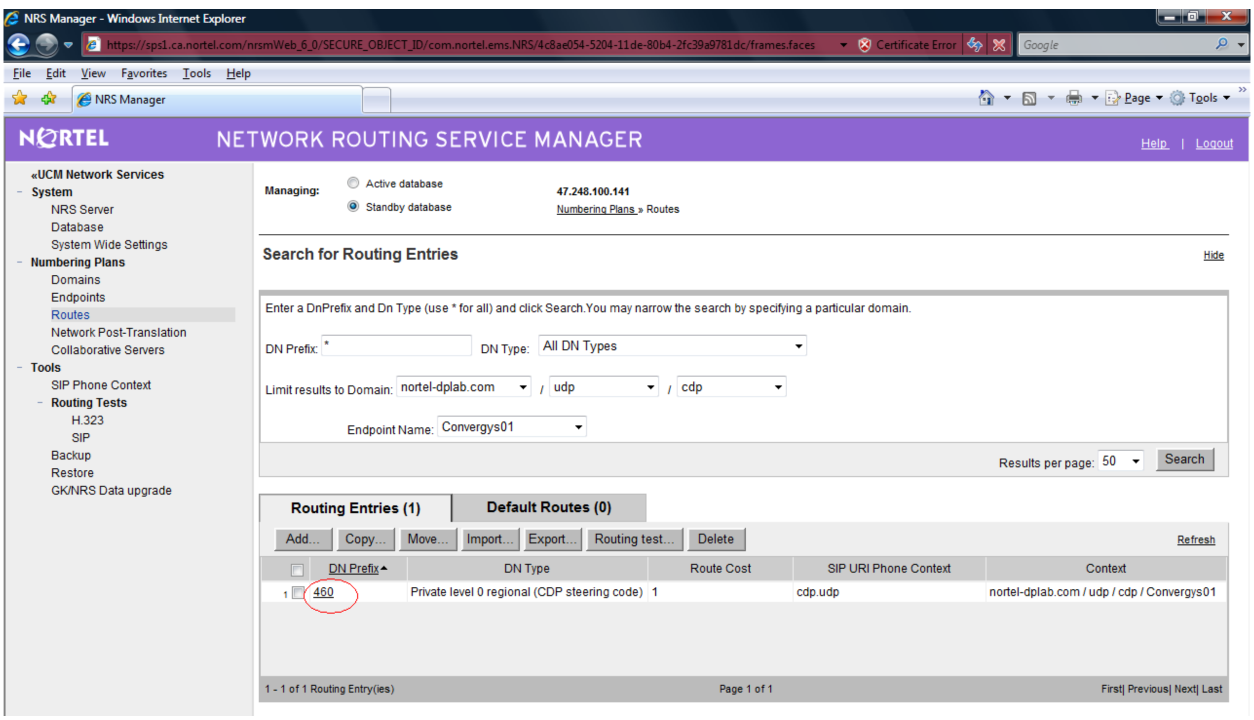

Figure 10: Route Entered for Convergys Endpoint

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 Convergys MS.

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.

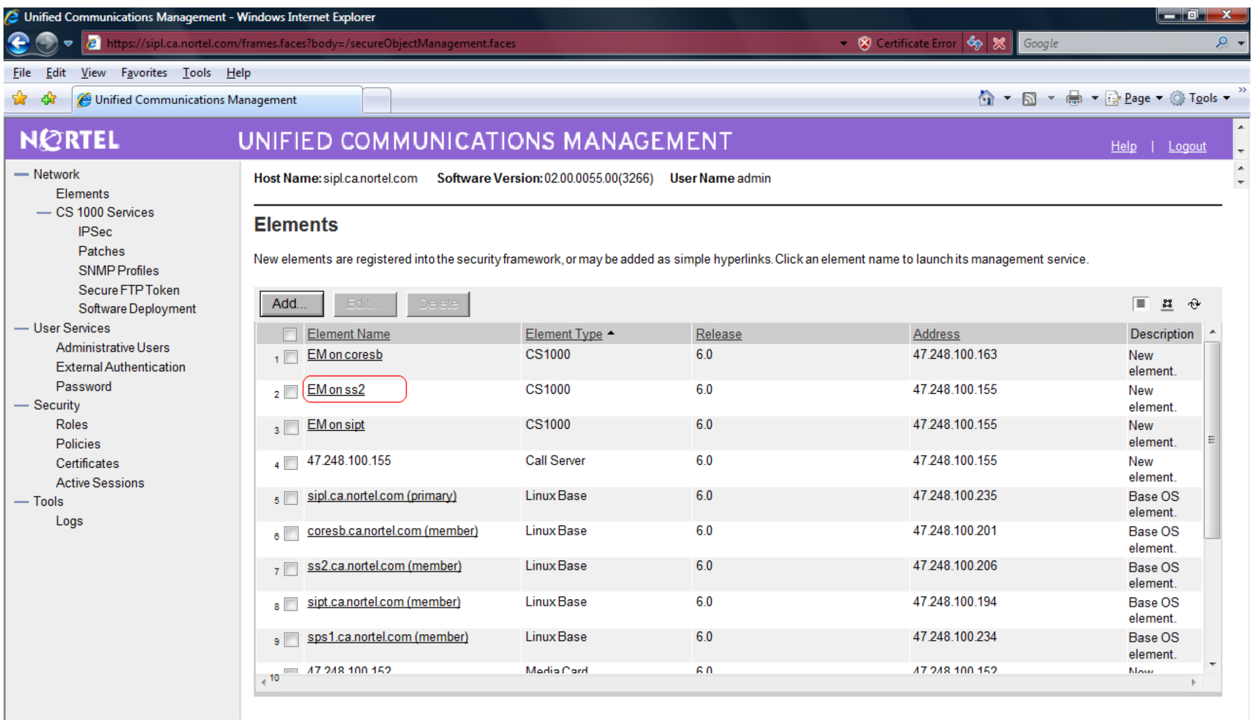

Figure 11: EM of Call Server on the UCM.

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

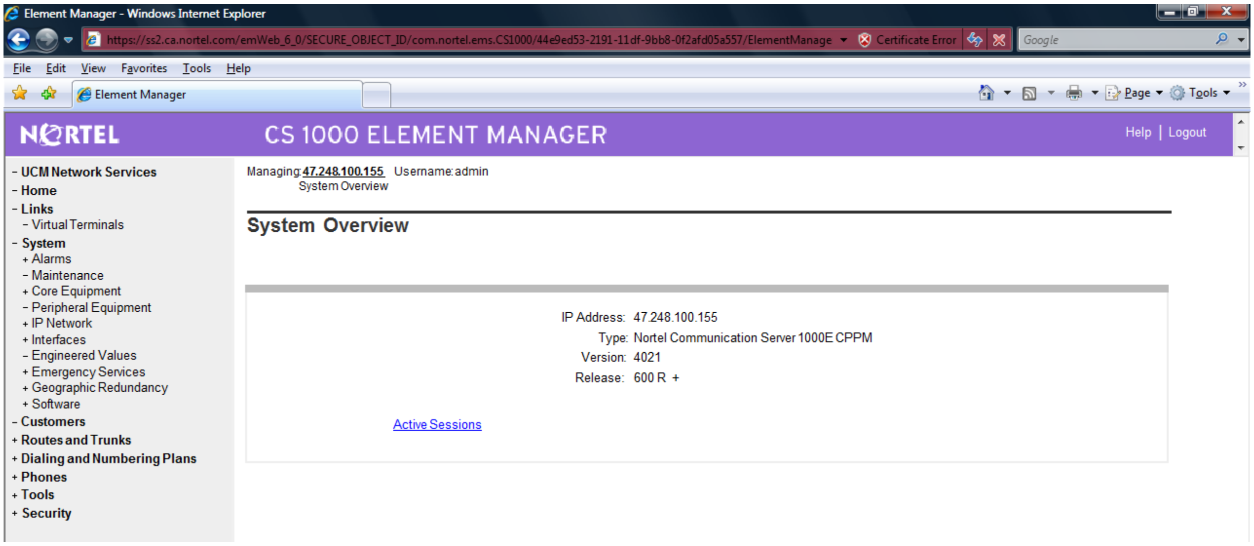

Figure 12: Element Manger homepage of Call Server

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.**

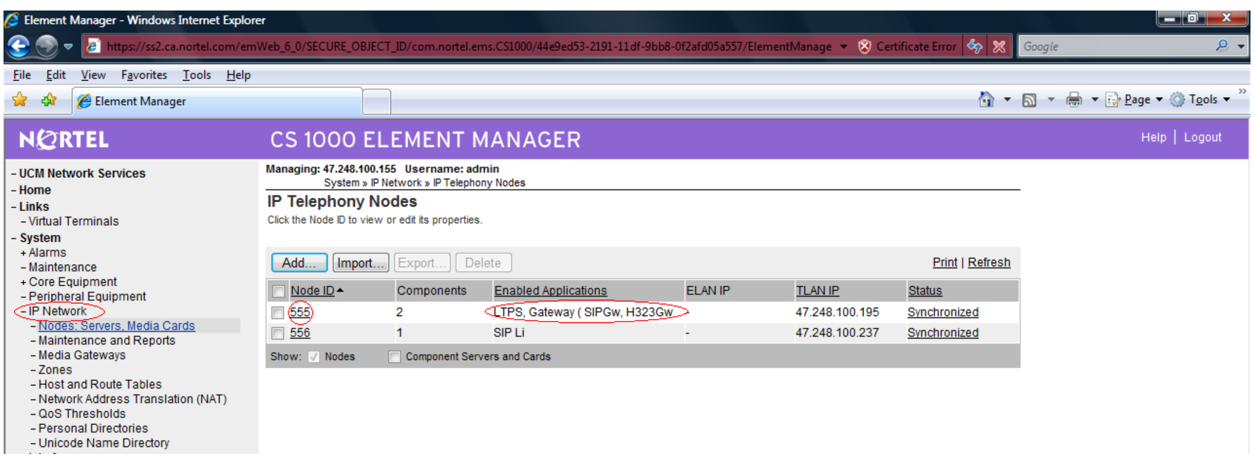

Figure 13: IP Telephone Nodes page of EM

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

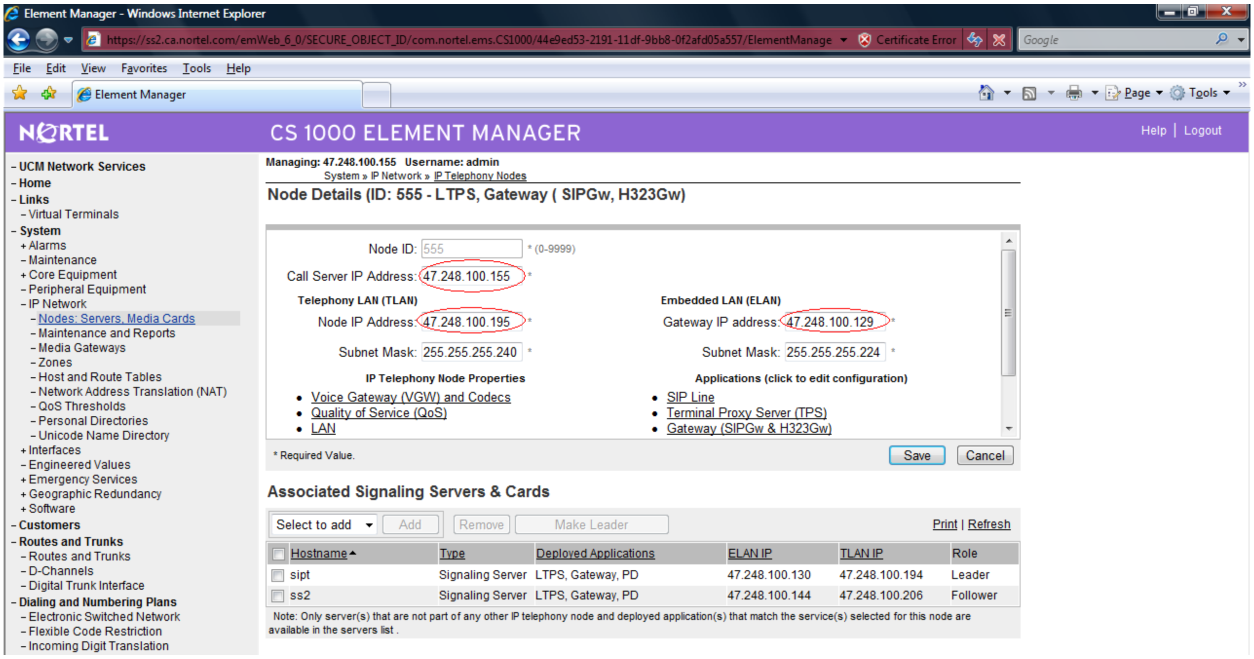

Figure 14: Node ID 555 page on the EM

- 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.

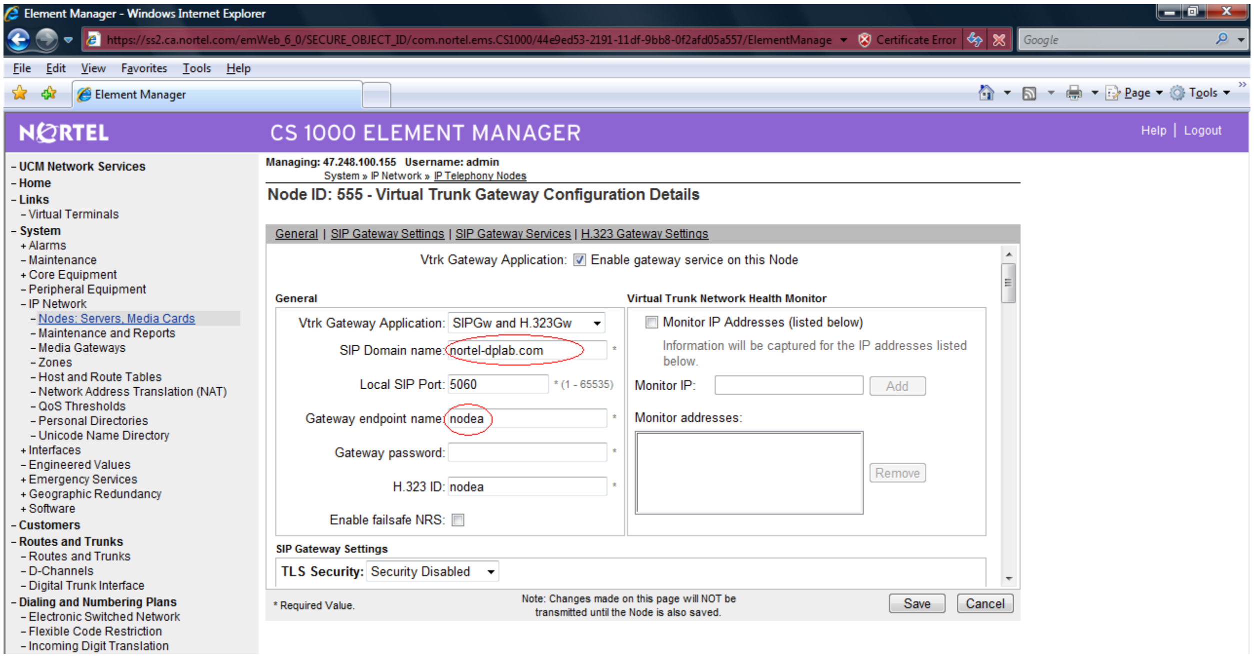

Figure 15: Virtual Trunk Gateway Configuration Details Page of EM

Under **the SIP Gateway Settings** section, the details configuration is filled out as shown in red circle, 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

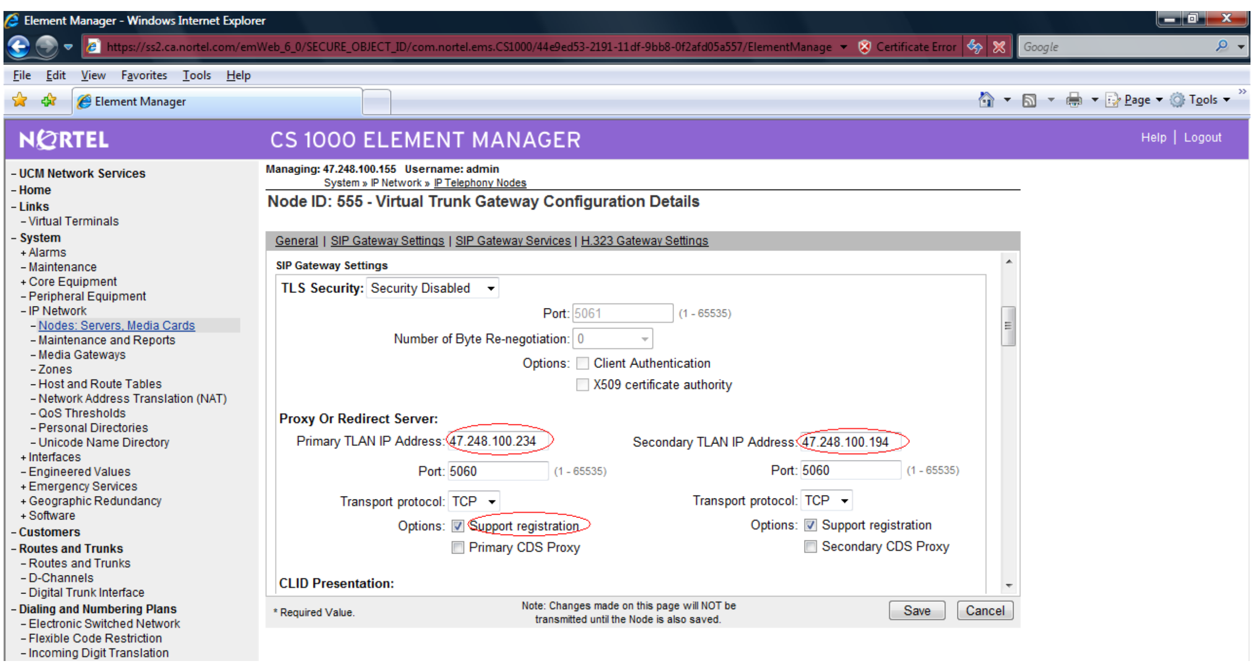

Figure 15b: Virtual Trunk Gateway Configuration Details page of EM

- 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 user to **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.

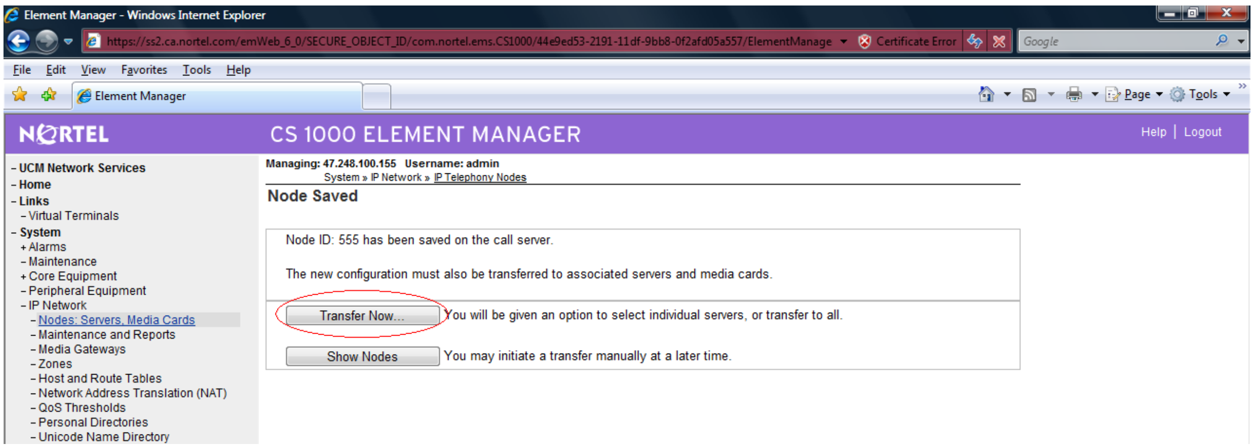

Figure 16: Transfer Now action of the Node Saved Page

<u>**Note**</u>: 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.

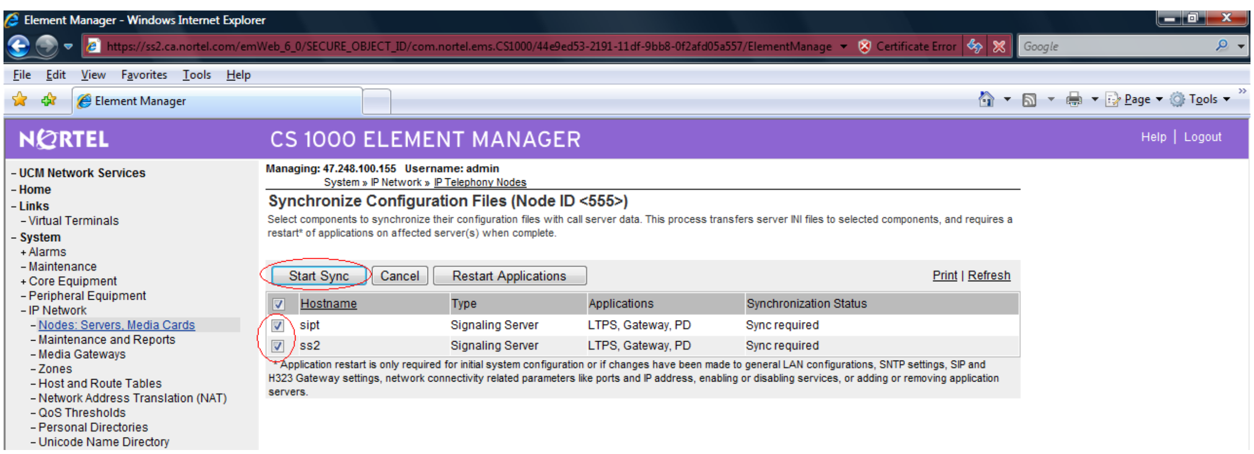

Figure 17 Synchronized Changes on the EM

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.

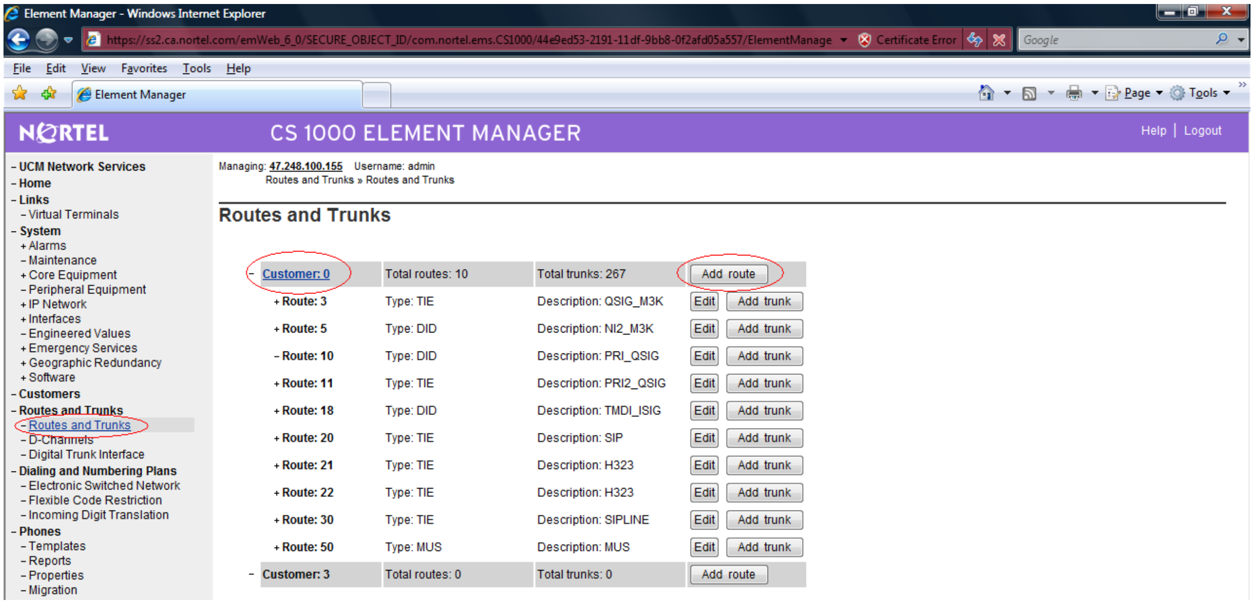

Figure 18: Routes and Trunks webpage

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***
- **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)***

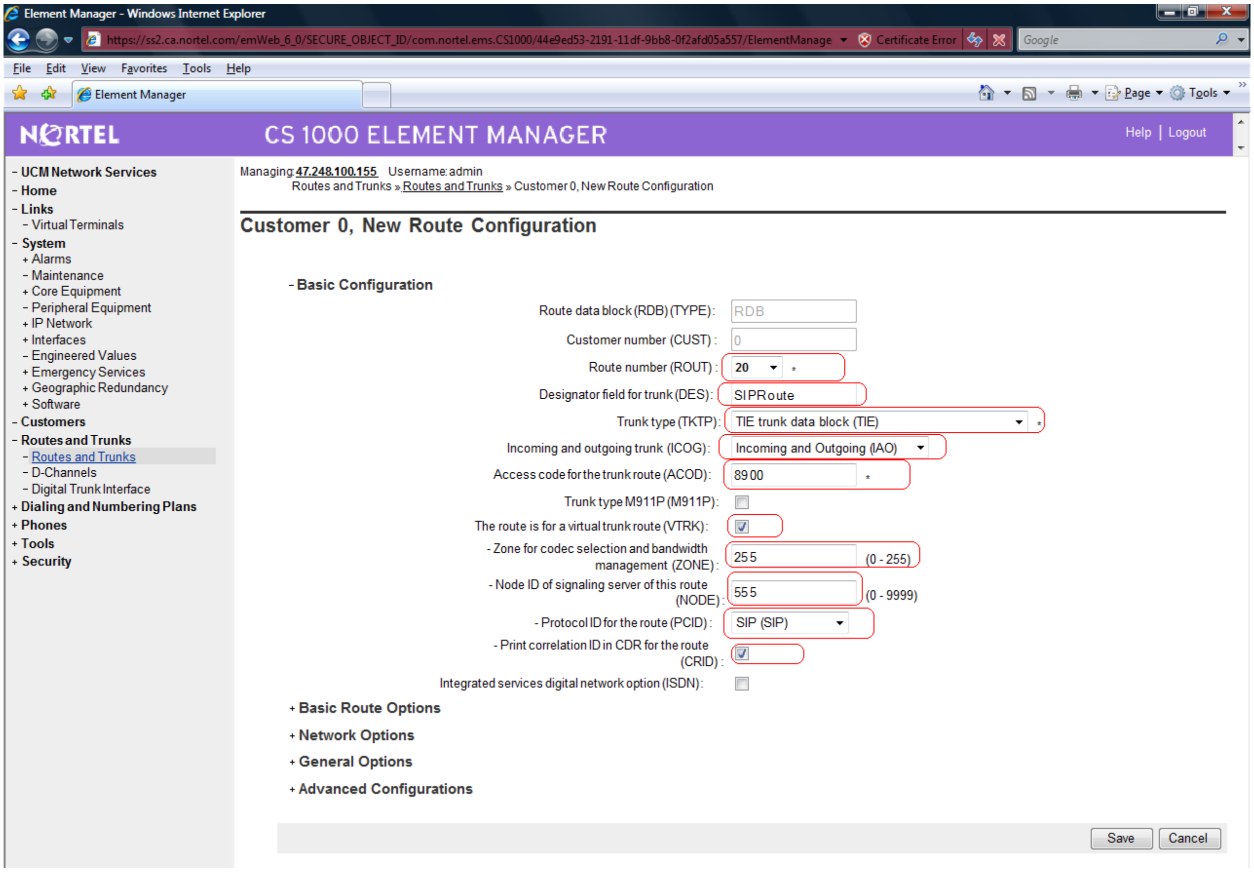

Figure 19: Basic Configuration of RDB

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

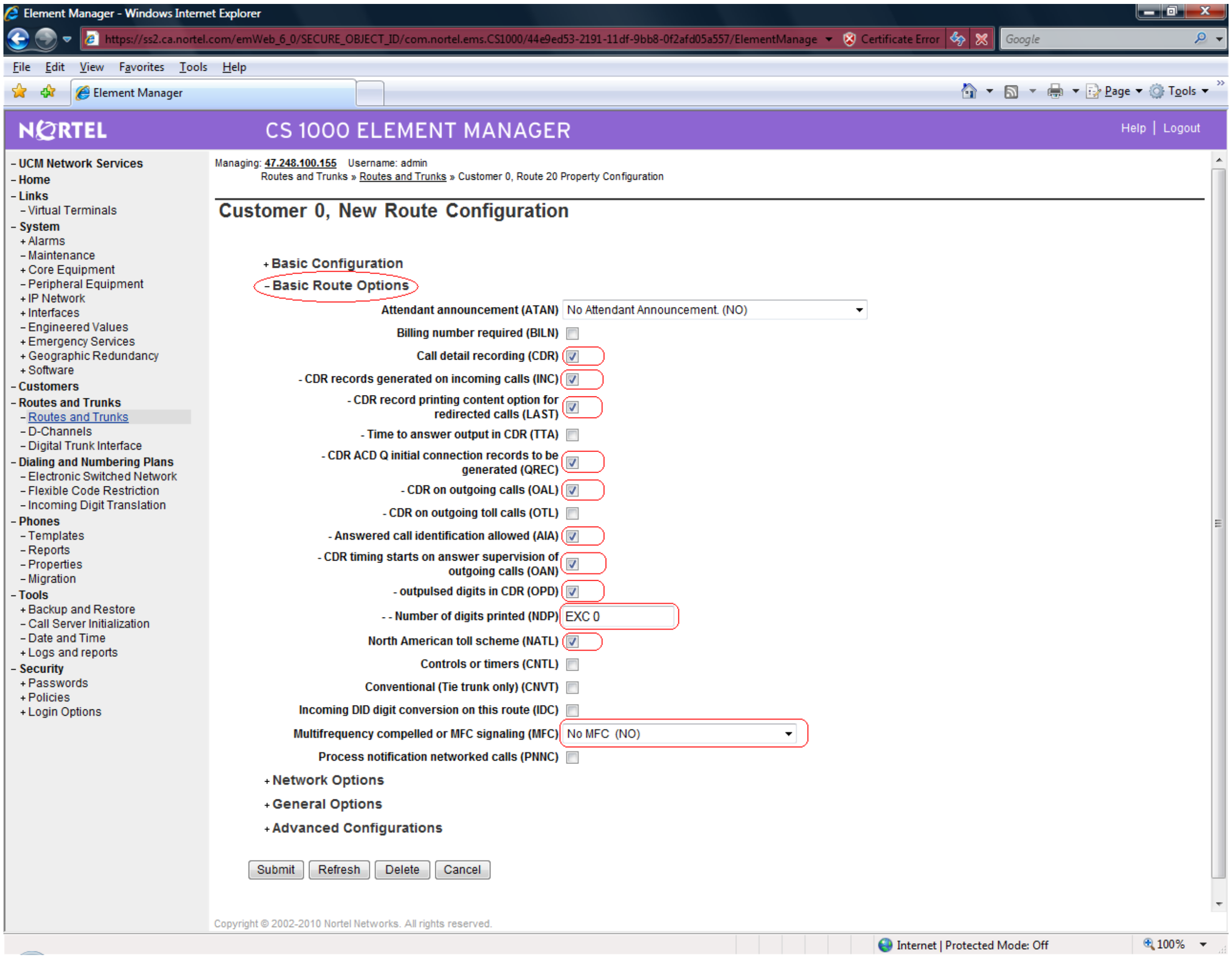

Figure 20: Basic Route Options of RDB

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

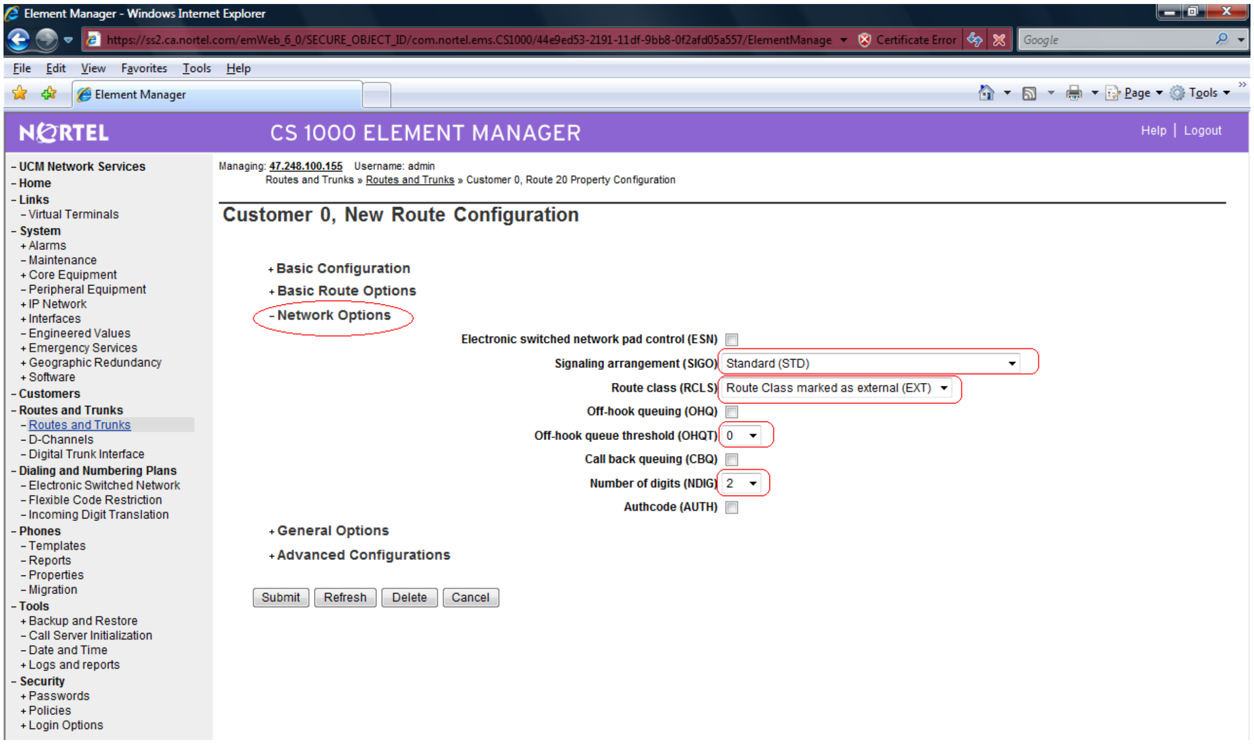

Figure 21: Network Options of RDB

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

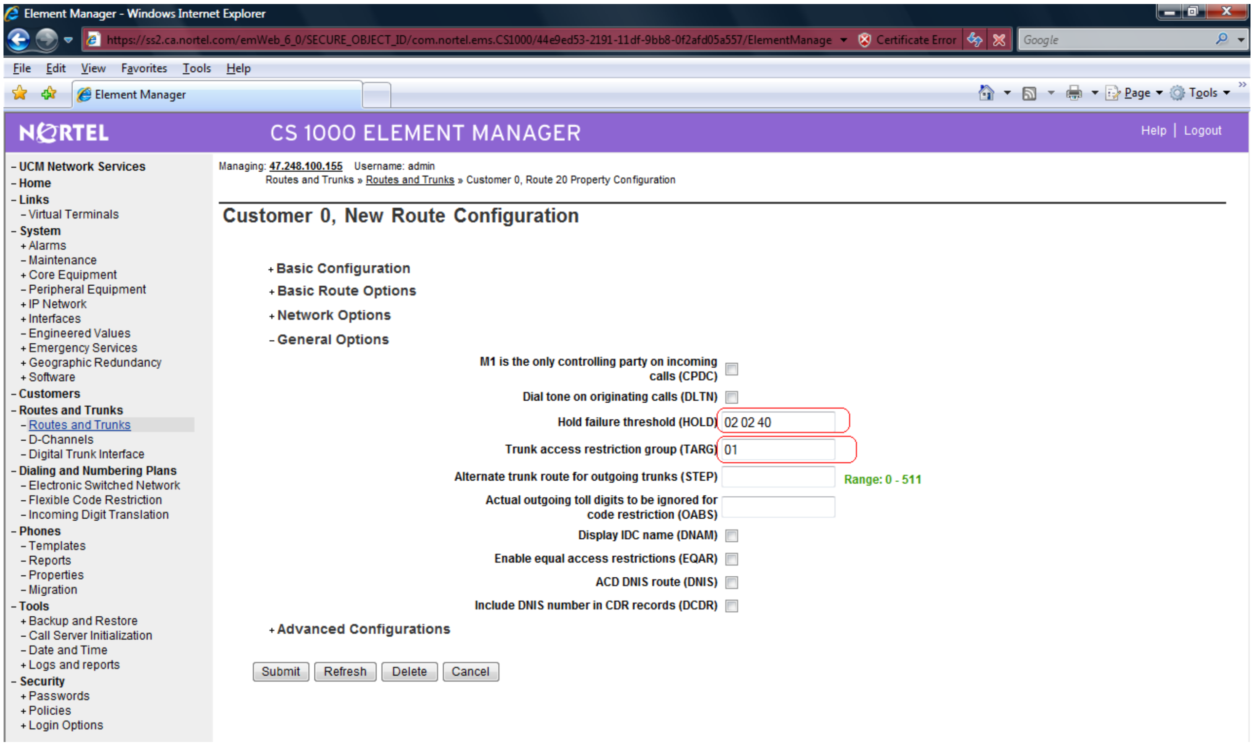

Figure 22: General Options Section of RDB

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

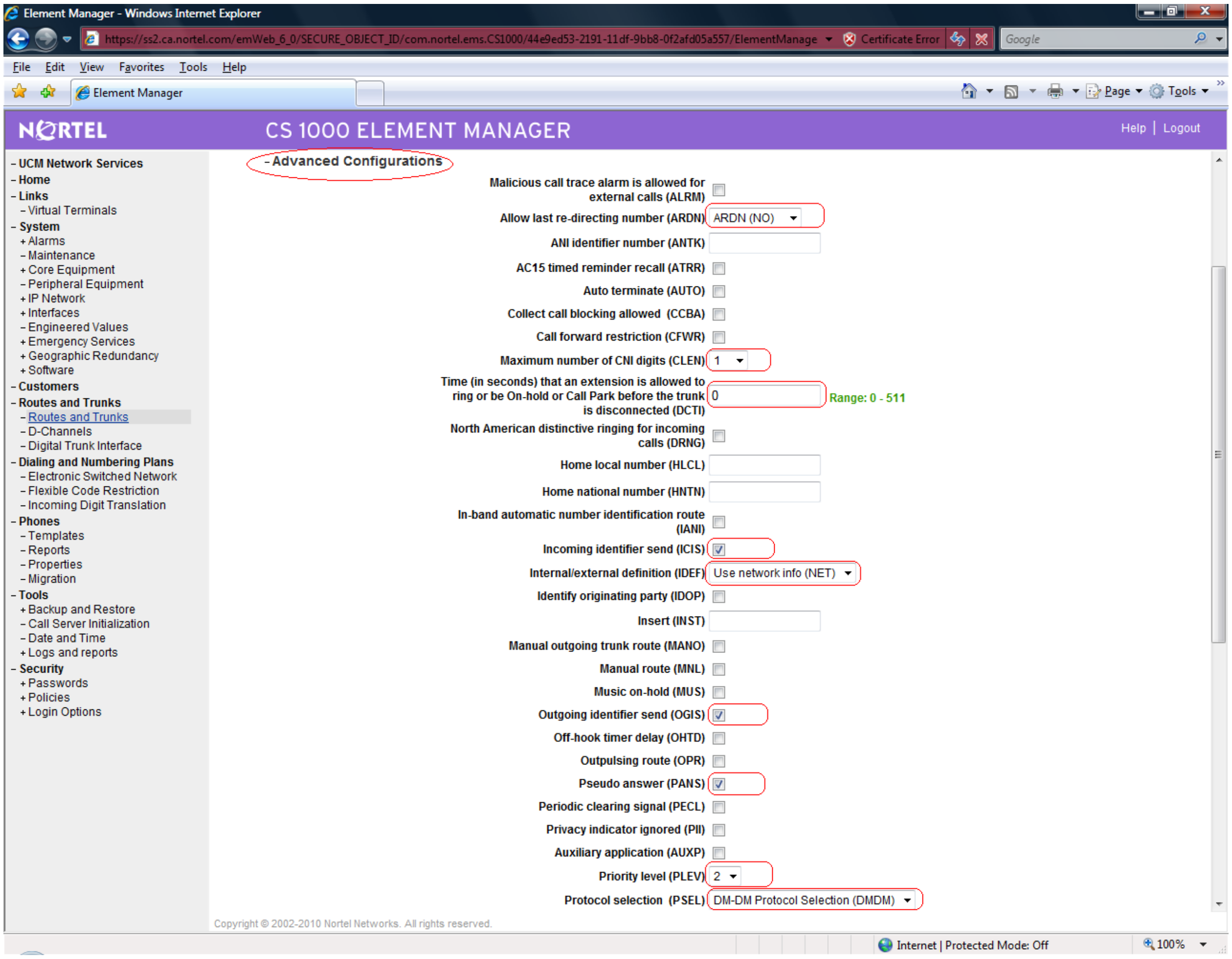

Figure 23: Advanced Configurations Section of RDB

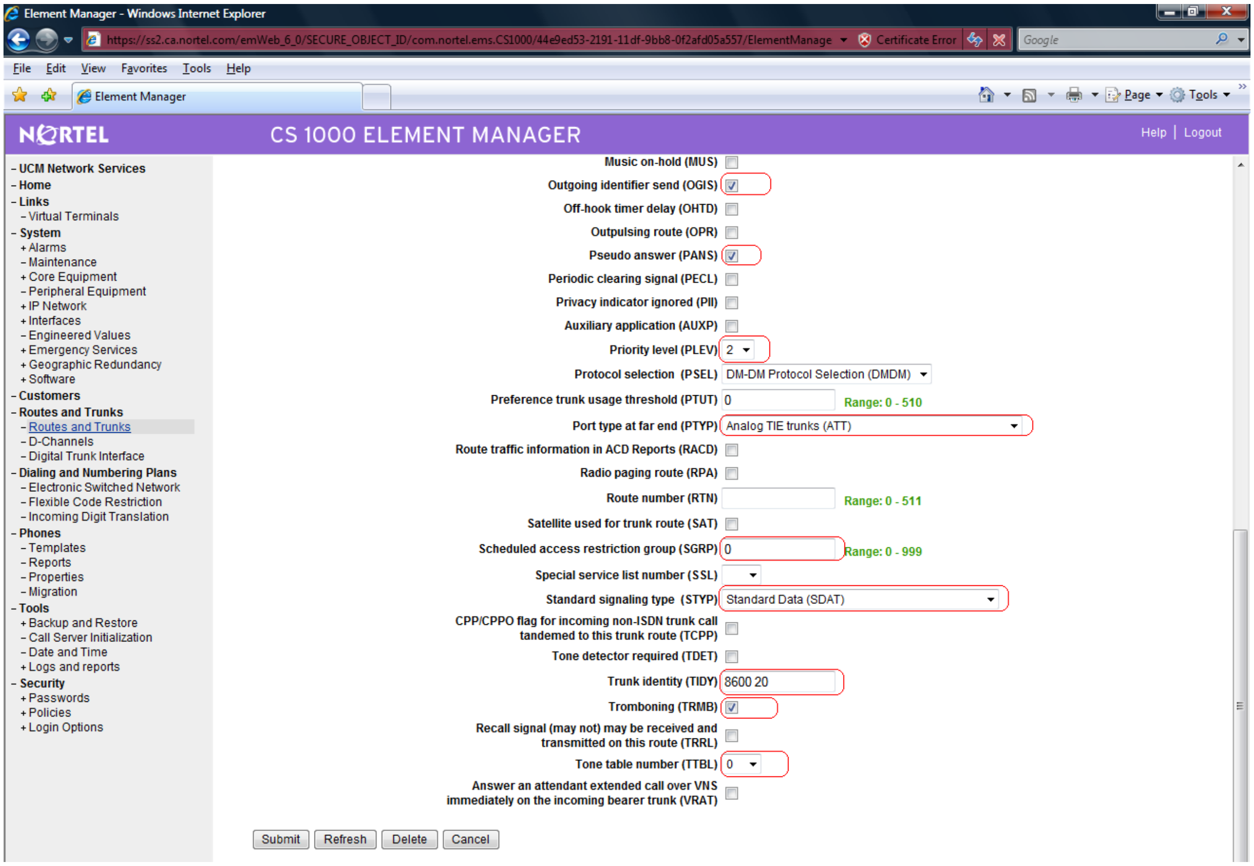

Figure 23b: Advanced Configurations Section of RDB (Continued)

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.

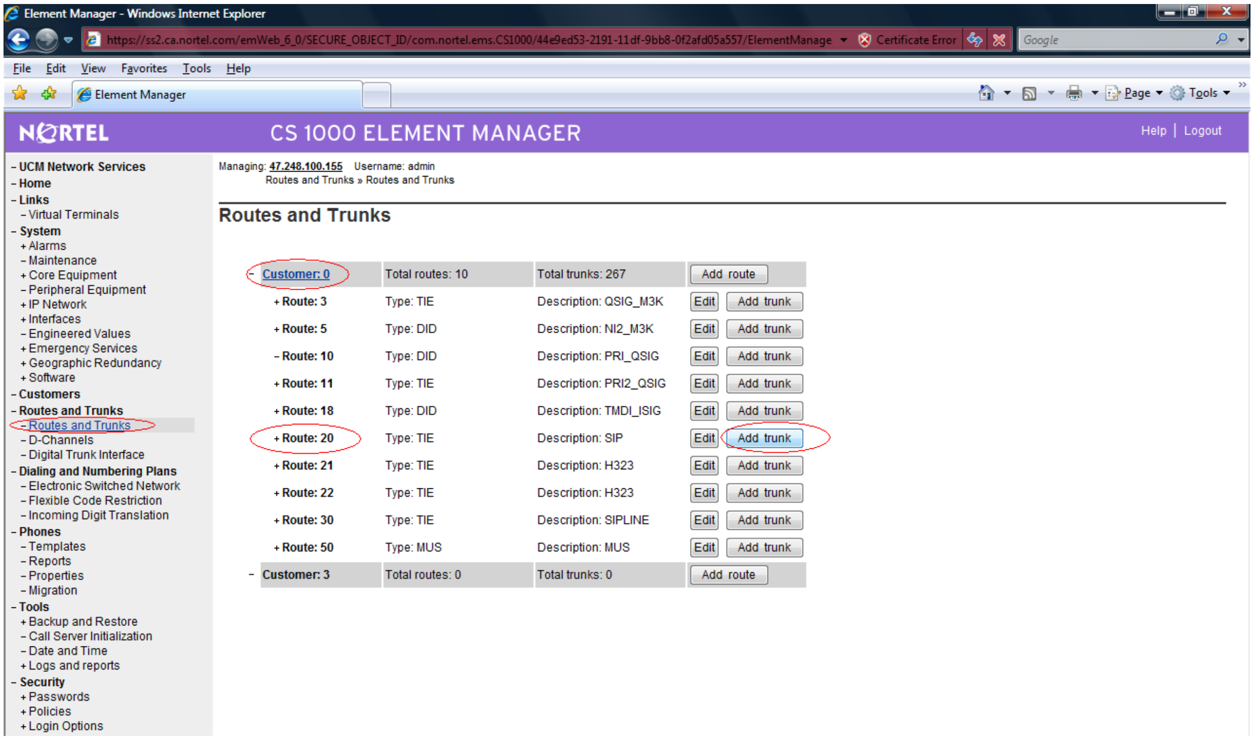

Figure 24: Routes and Trunks Properties Page of the EM

The **New Trunk Configuration** page will appear. Click on the **Basic Configuration**, the attributes below are populated as shown in red circle, 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).

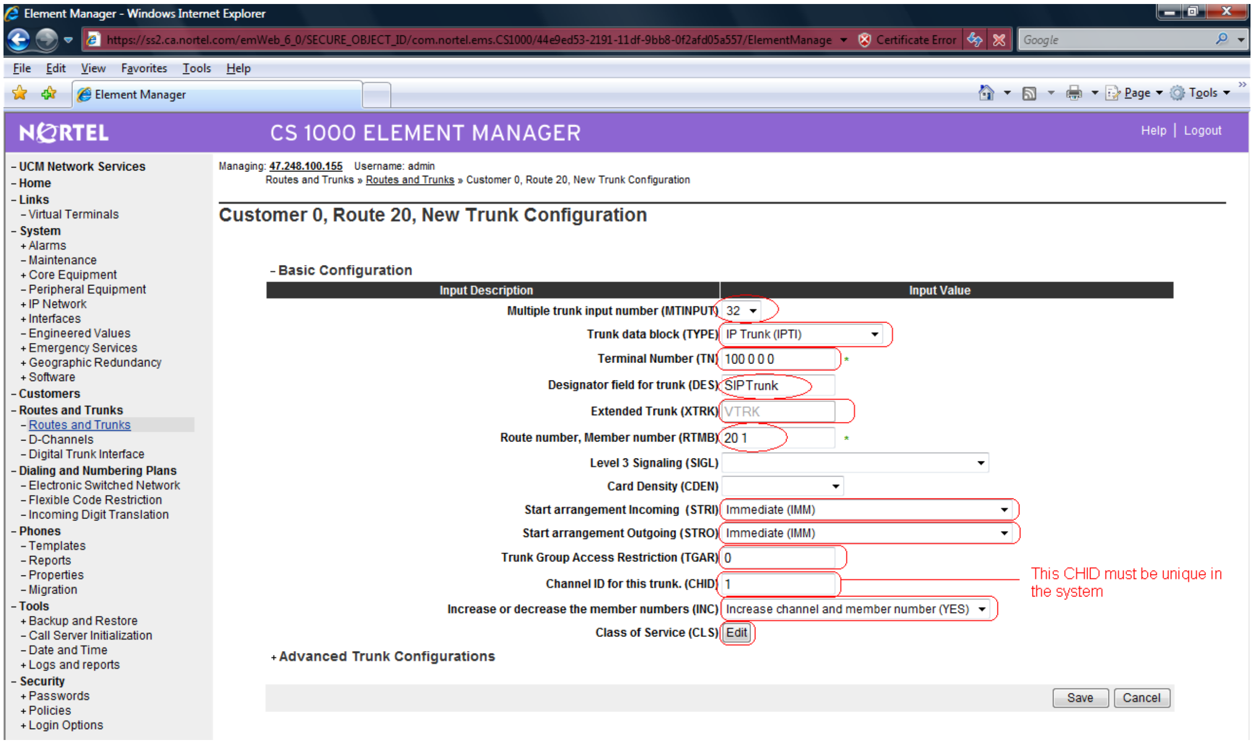

Figure 25: New Trunk Configuration

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

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

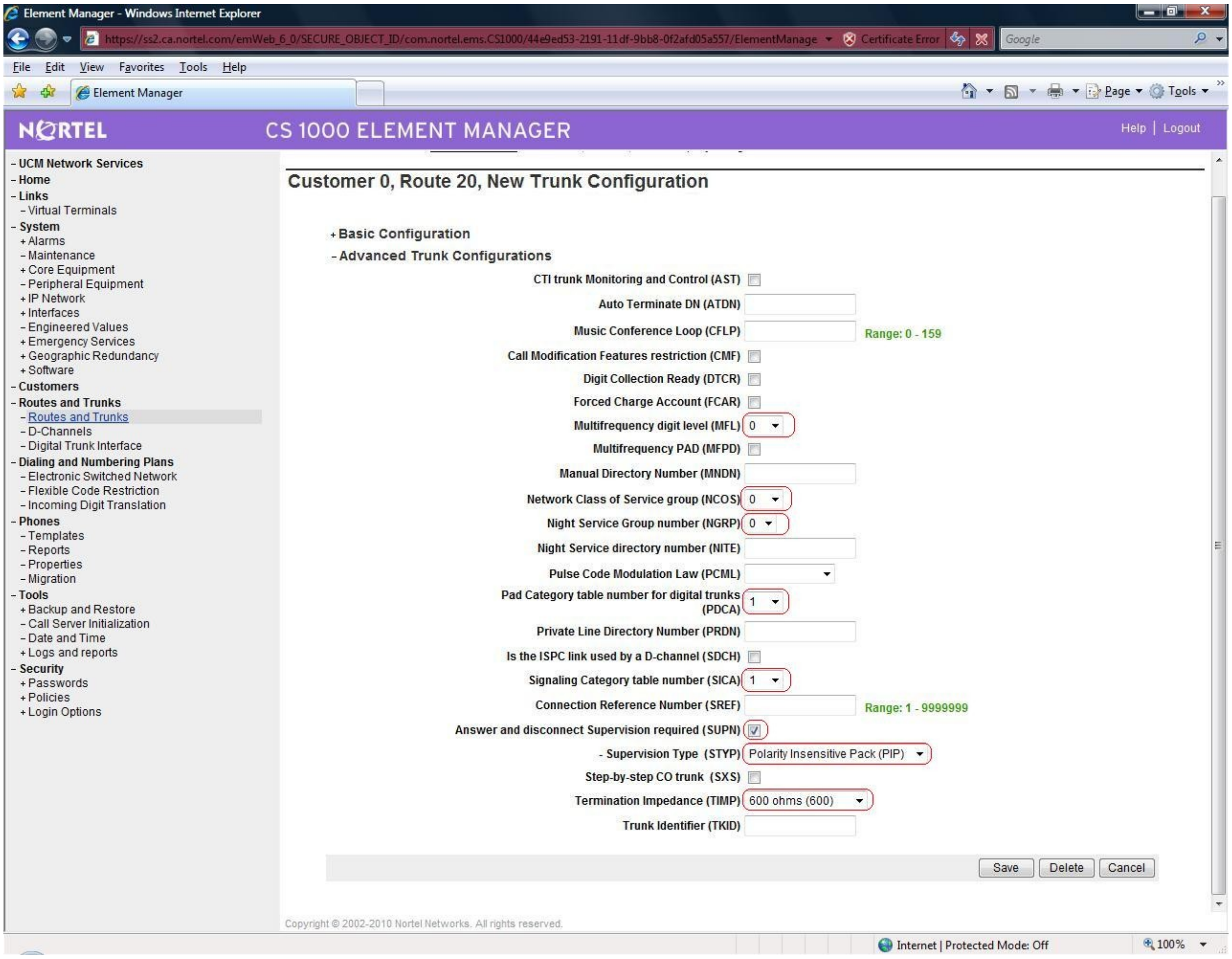

Figure 26: Advanced Trunk Configurations

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)

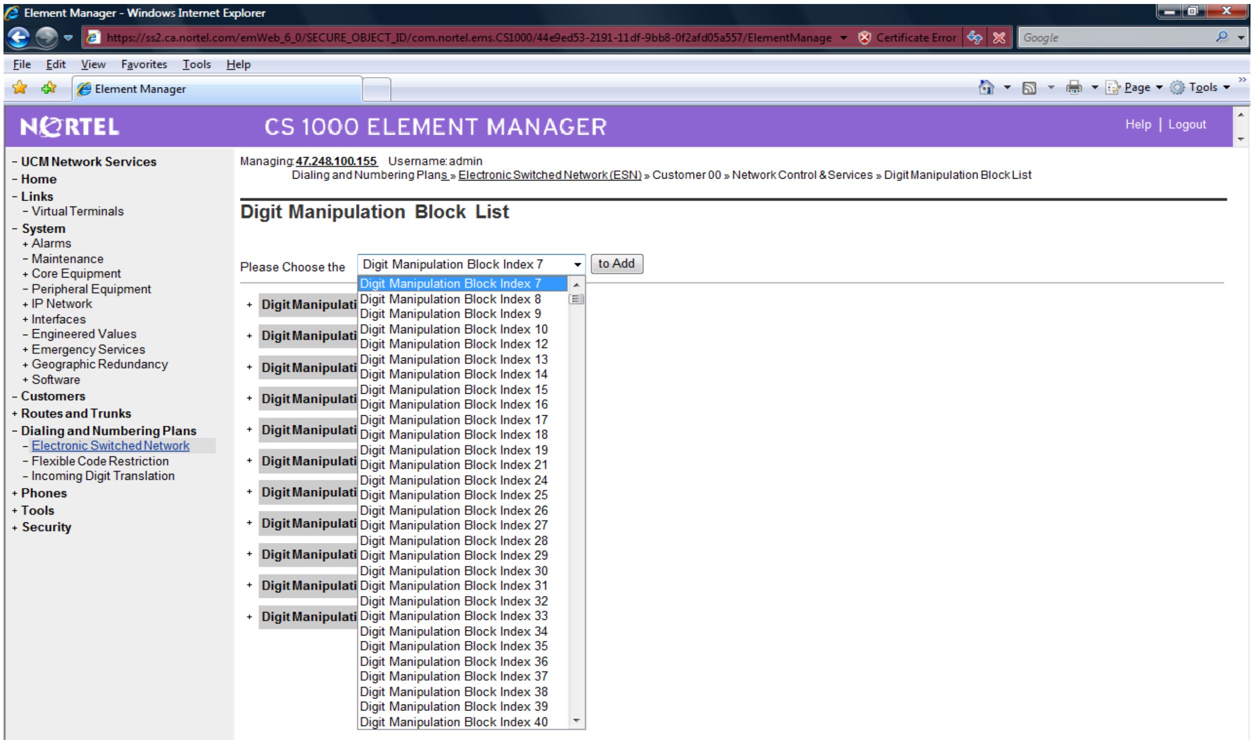

Figure 27: Digit Manipulation Block list

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.

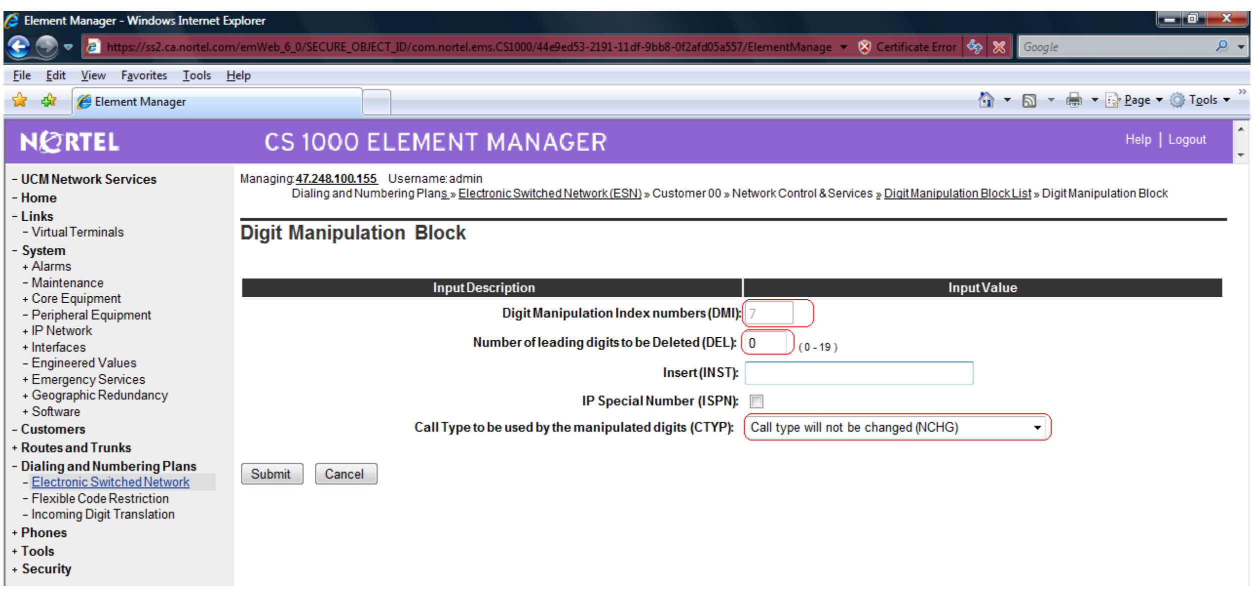

Figure 28: Digit Manipulation Block page

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.

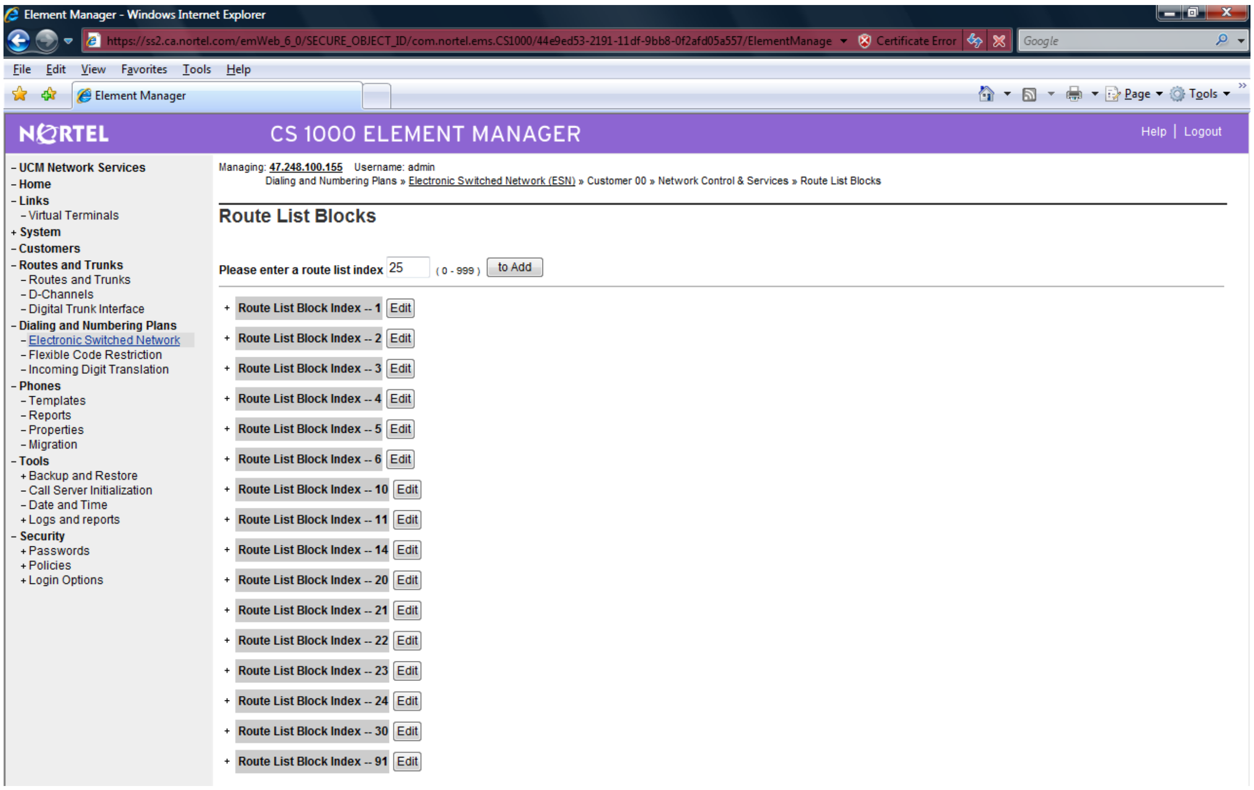

Figure 29: Route List Blocks page

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.

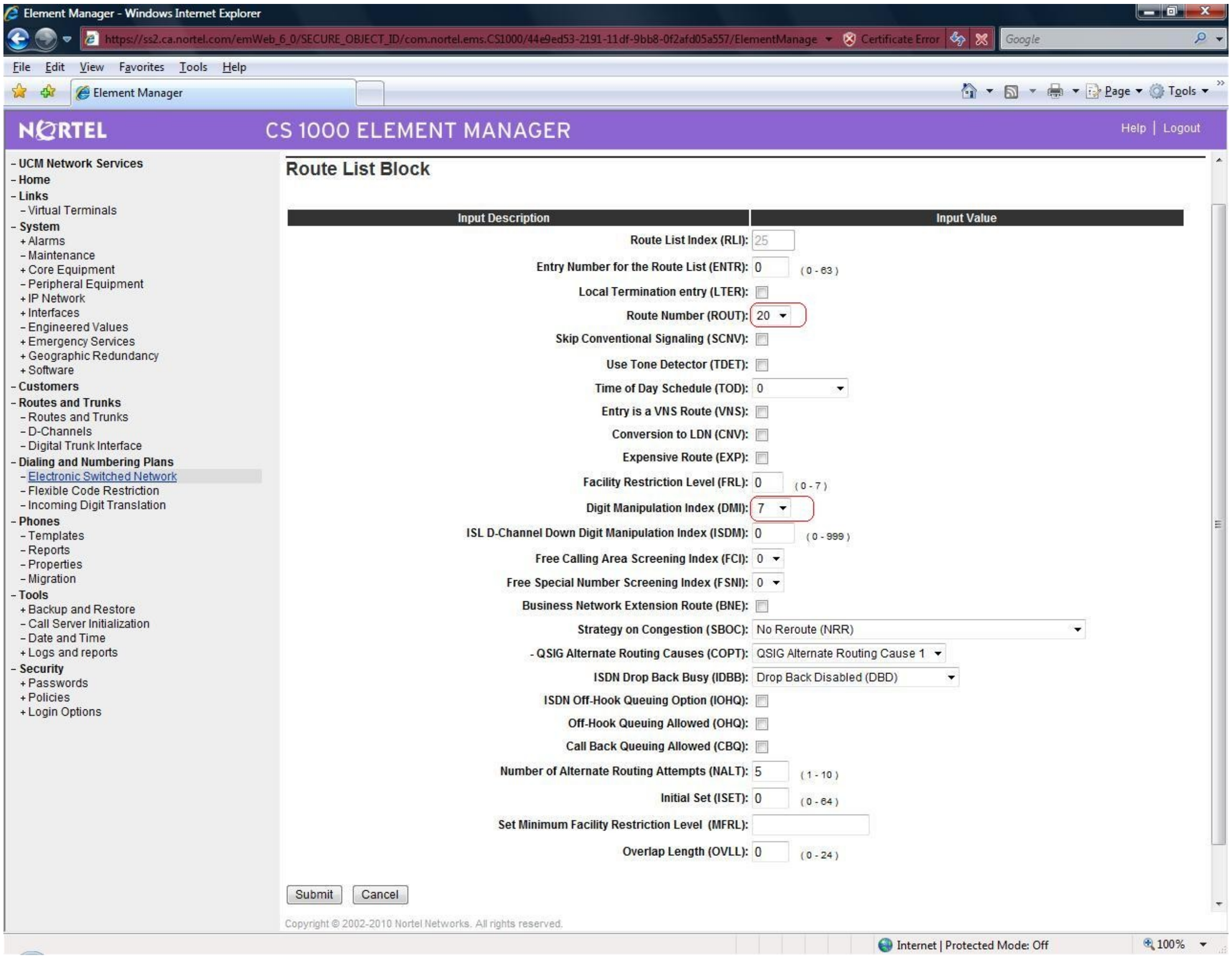

Figure 30: Detail of Route List Block page

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 **460** (in this example) which is the DN prefix of the Convergys MS that has been configured in section 4.2.2, Figure 9. Then click on the **to Add** button.

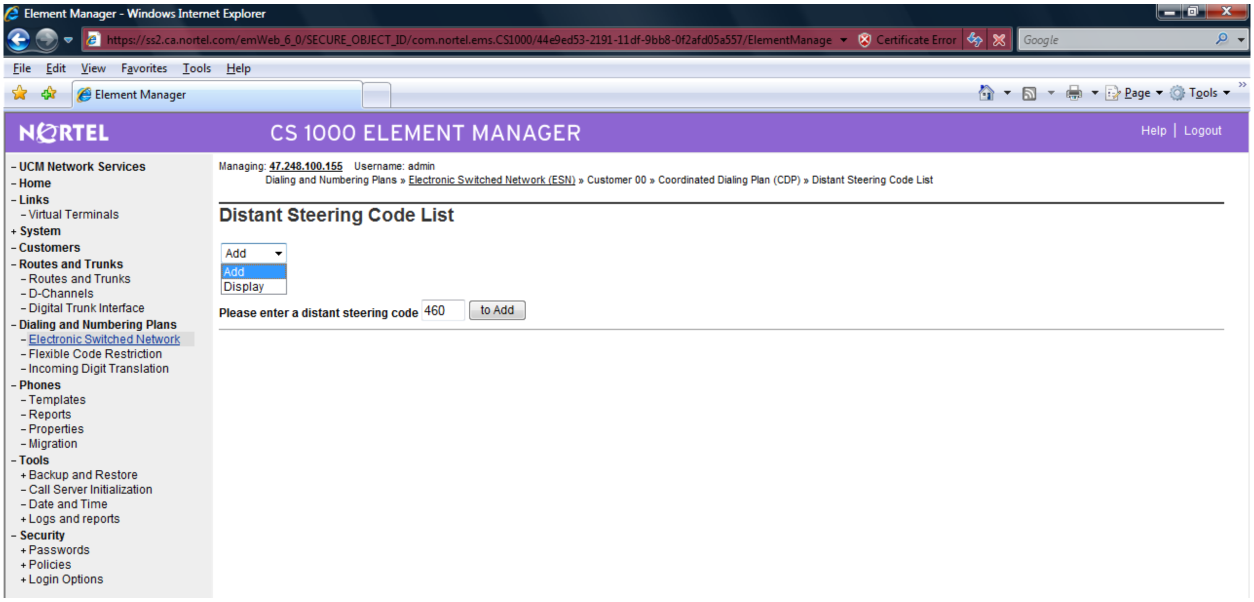

Figure 31: Distant Steering Code List page

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 access 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.

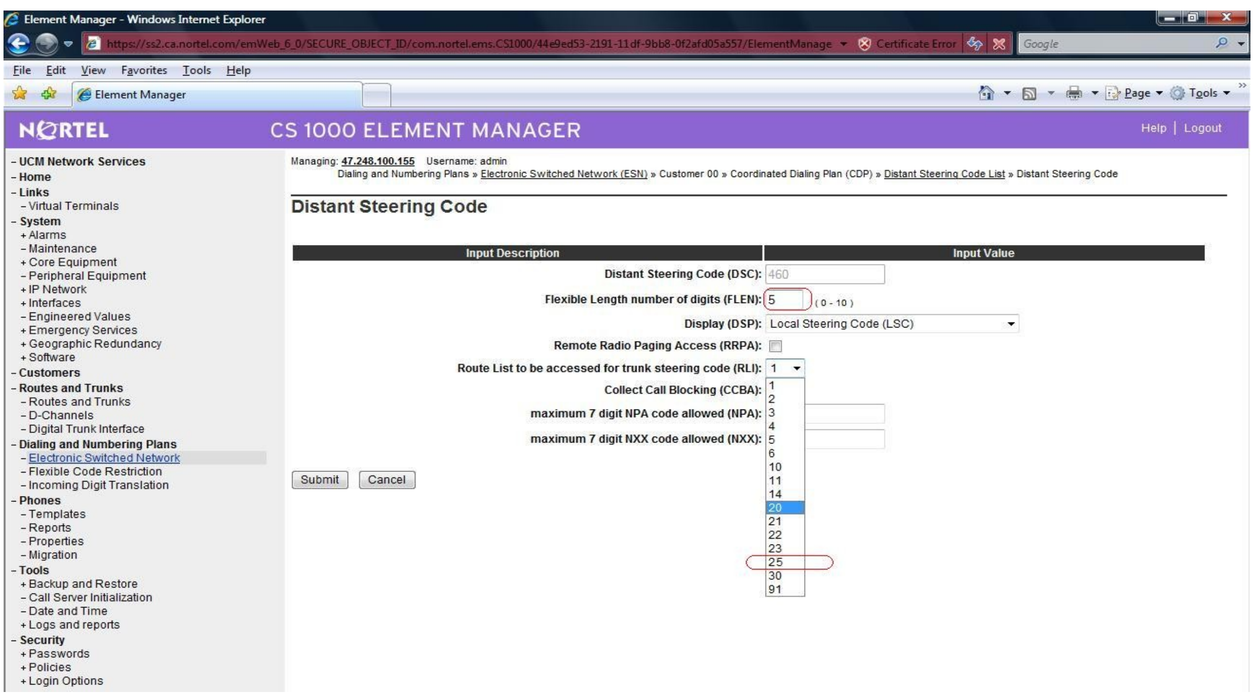

Figure 32: Detail of Distant Steering Code page

5. Configuring Convergys Intervoice Media Server 4.0

This section describes how to configure the Convergys MS to inter-work with the CS 1000 system.

5.1. Configuring Convergys HAL Boards

The Intel HMP software currently supports one Intel HMP device (virtual board). Therefore, even though the Hardware Abstraction Layer (HLA) configuration allows for more than one device (board), only configure one device.

Open the Control Centre of the Convergys MS by addressing the IP address and port number of the Convergys MS in the Microsoft Internet browser, https://47.248.100.143:8074/ccportal, and then input the user name and password to login to the **Control Center**, click on the **Accept** button (not shown).

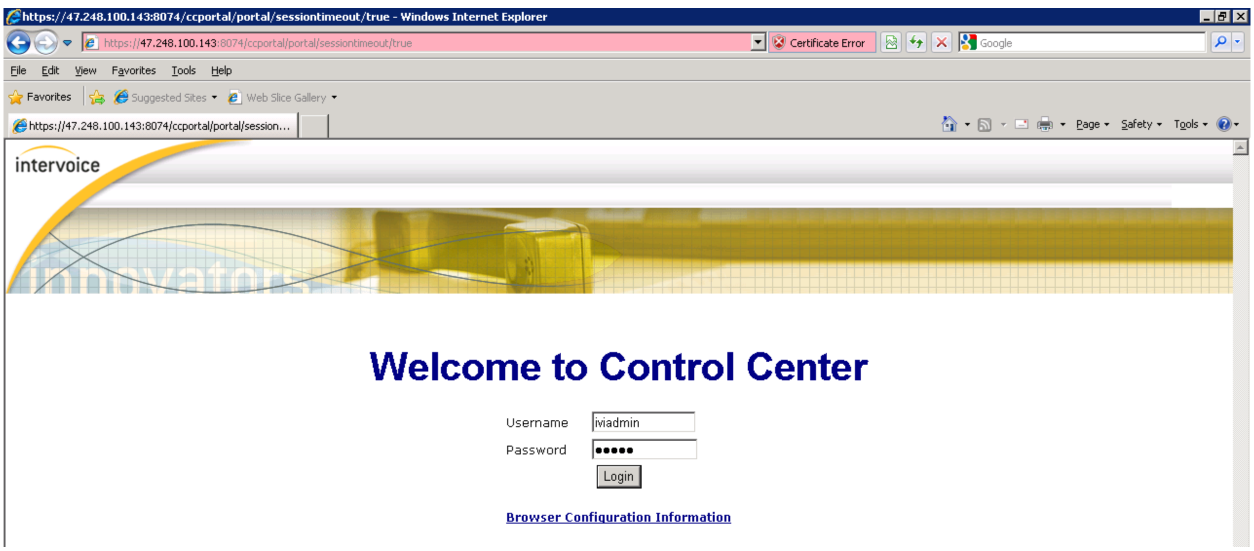

Figure 33: Control Center homepage

To configure the HAL board settings, navigate to the **Node view → Configure → Configurations → Unassigned → HMP IVR (47.248.100.143) -> Media Server 3.5 and above →HAL HMP Configurations → View/Edit Boards** (not shown). Click on the **Edit** link of the **Board ID** 0**,** the **Board** details page will appear as shown in Figure 34. Enter the **IP address** 47.248.100.143 (the IP address of Intel HMP device) and the **Protocol Name** as SIP. Click on the Submit button to save the configuration changes.

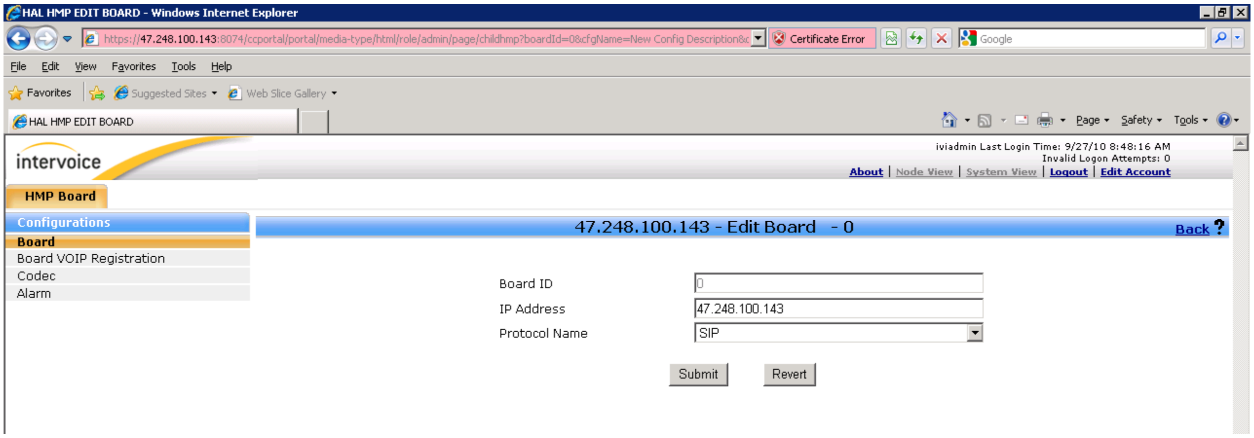

Figure 34: HMP board configuration page

To configure the **Board VOIP Registration**, navigate to the **Node view → Configure → Configurations → Unassigned → HMP IVR (47.248.100.143) -> Media Server 3.5 and above →HAL HMP Configurations → View/Edit Boards** (not shown). Click on the **Edit** link and select the **Board VOIP Registration, the Board VOIP Registration** page will appear as shown in Figure 35. Click on the **Add** link under **Actions** attribute to add a SIP domain URI entry for the registration to PBX gateway/proxy (NRS in this example). In this example, the **Alias String**: Convergys01@nortel-dplab.com is used as the URI address which has the domain name, nortel-dplab.com, of the CS 1000 NRS.
Click on the **Submit** button to complete the configuration changes.
Note: The VoIP protocol supports only the e-mail alias type and can have one alias string defined.

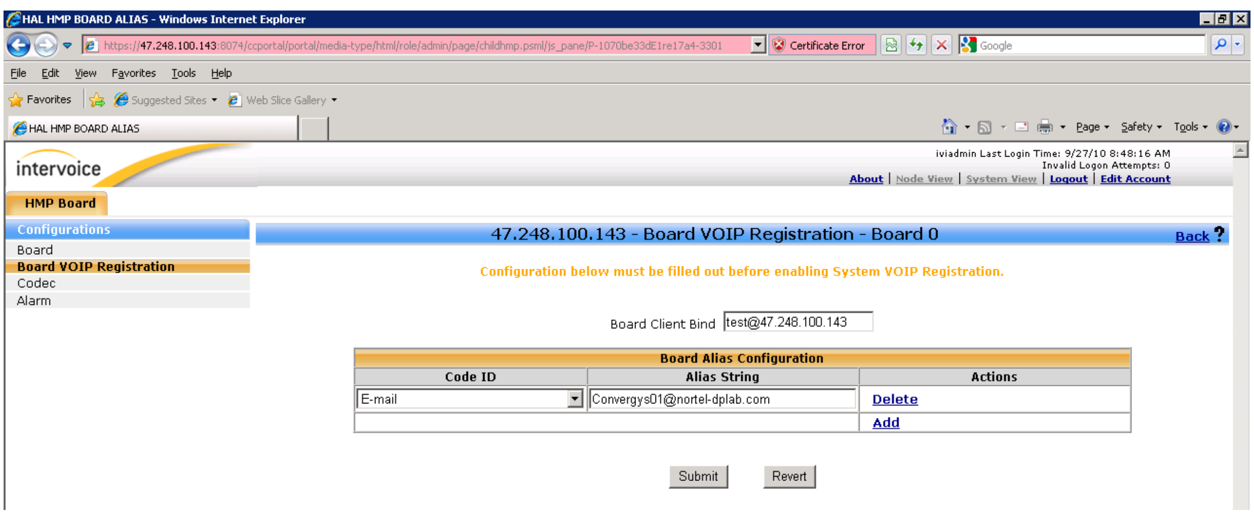

Figure 35: Board VOIP Registration Configuration

5.2. Enabling System VOIP Registration

To configure the **Board VOIP Registration**, navigate to the **Node view → Configure → Configurations → Unassigned → HMP IVR (47.248.100.143) -> Media Server 3.5 and above →HAL HMP Configurations → System VOIP Parameters → System VOIP Registration**, the **System VOIP Registration** details page will appear as shown in figure 36.

Click on the **Add** link under **Actions** attribute to add the address of the SIP gateway/Proxy of the PBX IP (CS 1000 in this case) which the Convergys MS is going to register to. In this example the **VOIP Registry Address** is 47.248.100.234, the **Hops Count** is 100, the **Registration Frequency** is PERIODIC, the **Registration Interval (ms)** is 300000 and the **Enable VOIP Registration** checkbox is checked. Click on the **Submit** button to complete the configuration changes as shown in figure 36.

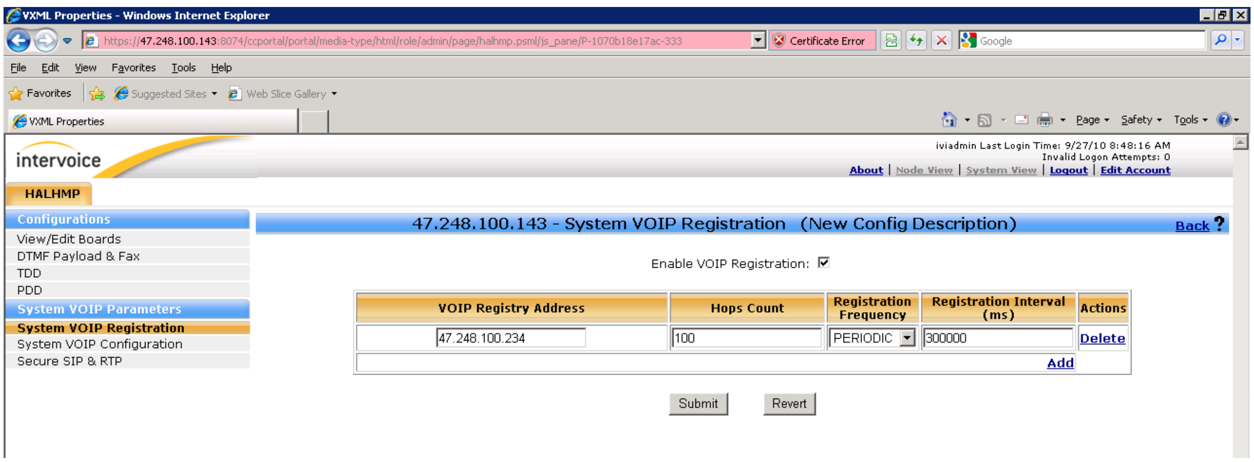

Figure 36: System VOIP Registration

To configure the **Board VOIP Registration**, navigate to the **Node view → Configure → Configurations → Unassigned → HMP IVR (47.248.100.143) -> Media Server 3.5 and above →HAL HMP Configurations → System VOIP Parameters → System VOIP Configuration**, the **System VOIP Configuration** details page will appear as shown in Figure 37. Check the **Enable TCP** checkbox to use the TCP protocol and uncheck to use UDP when sending an outgoing call.

Note: Incoming UDP and TCP messages are accepted, regardless of the setting of this field.

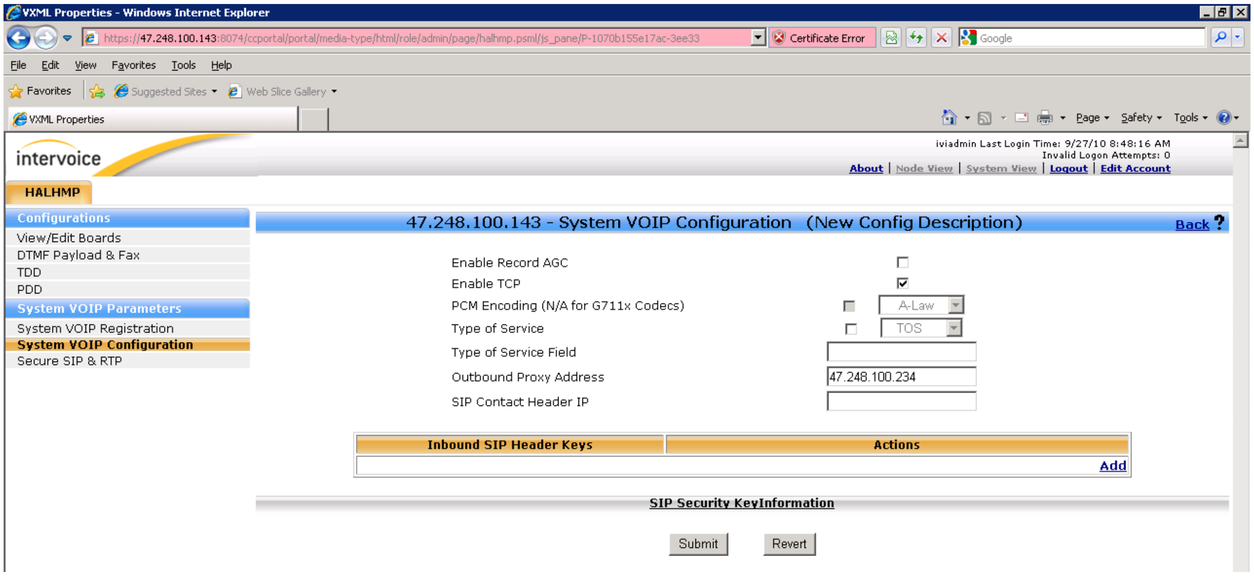

Figure 37: System VOIP Configuration

5.3. Configuring Codec

To configure codec use on the Convergys MS, navigate to the **Node view → Configure → Configurations → Unassigned → HMP IVR (47.248.100.143) -> Media Server 3.5 and above →HAL HMP Configurations → View/Edit Boards** (not shown). Click on the **Edit** link and select the **Codec**, the Codec page details will appear as shown in Figure 38. Click on the **Add** link to add codec, its frame size and frame per packet to be used. After choosing the appropriate codec to be used, click on the **Submit** button to save the changes.

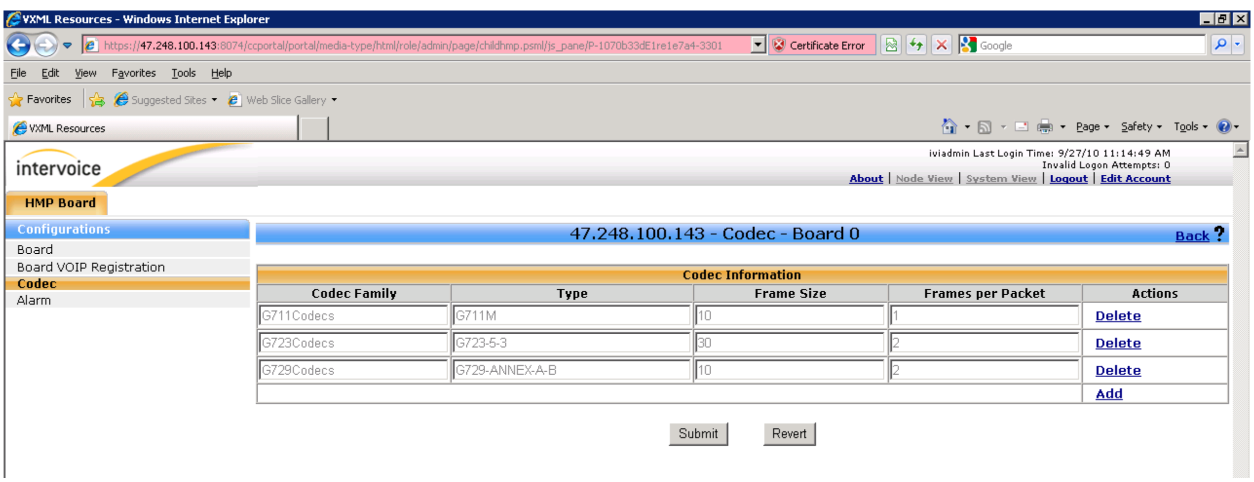

Figure 38: Codec Board Configuration

Note: Any changes on the Convergys MS need a restart the Intervoice Media Server service of Convergys MS to make it active. The Intervoice Media Server service can be restarted in the Windows Services application.

5.4. Configuring DTMF Payload and Fax

To configure DTMF Payload and Fax use on the Convergys MS, navigate to the **Node view → Configure → Configurations → Unassigned → HMP IVR (47.248.100.143) -> Media Server 3.5 and above →HAL HMP Configurations → DTMF Payload & Fax,** select the DTMF to be used and then click on the **Submit** button to save configuration changes as shown in Figure 39.

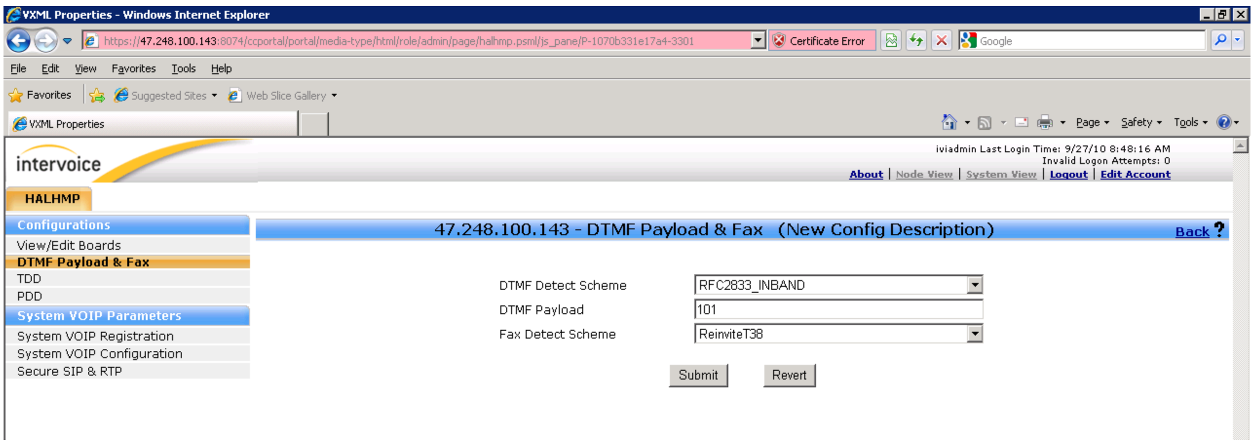

Figure 39: DTMF Payload and Fax Configuration

5.5. Configuring Call Control

To access and configure Call Control, navigate to the **Node view → Configure** (tab) **→ Configurations → Unassigned → HMP IVR (47.248.100.143) → Media Server 3.5 or above →Call Control Configurations (not shown),** click on the **Edit** link on the right menu column, **Actions**. The **Telephony – New Config Description** details page will appear as shown in Figure 40. Choose **VoIP** option from the pull down menu **Server Type**. Then click on the **Submit** button to save the configuration changes.

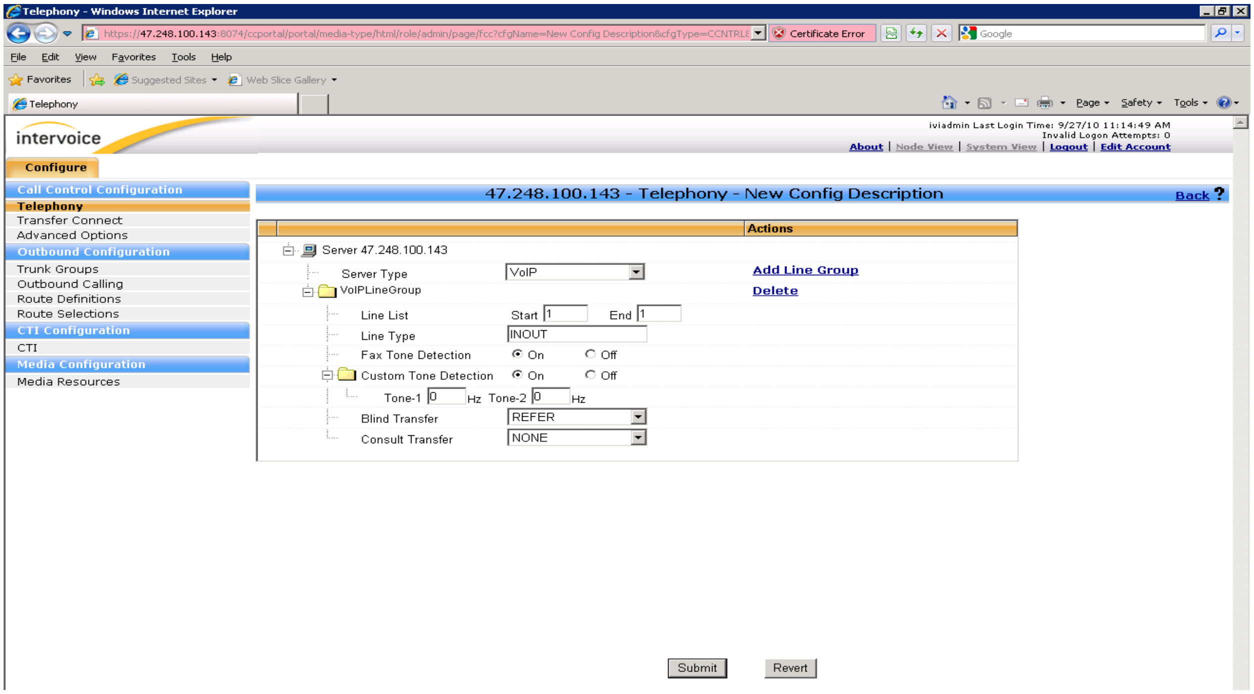

Figure 40: Telephony Configuration Page

5.6. Configuring Application Routing

To access and configure the Application Routing, navigate to the **Node view → Configure** (tab) **→ Configurations → Unassigned → HMP IVR (47.248.100.143) → Media Server 3.5 or above →Application Routing (not shown),** click on the **Edit** link on the right menu column, **Actions,** the **Configure** page will appears as shown in Figure 41.

Click on the **Add** button under the **Actions** attribute of the **Applications** menu to add a new application. In this example, **Application Name** is DefaultApplicationName, **Application Method** is get, **Application Type** is application/voicexml+xml, and **Application URL** is file://D:/Program Files/Intervoice/Media Server for VoIP/APPL/testapp_ulaw/testApp.vxm. Click on the **Submit** button to save the configuration.

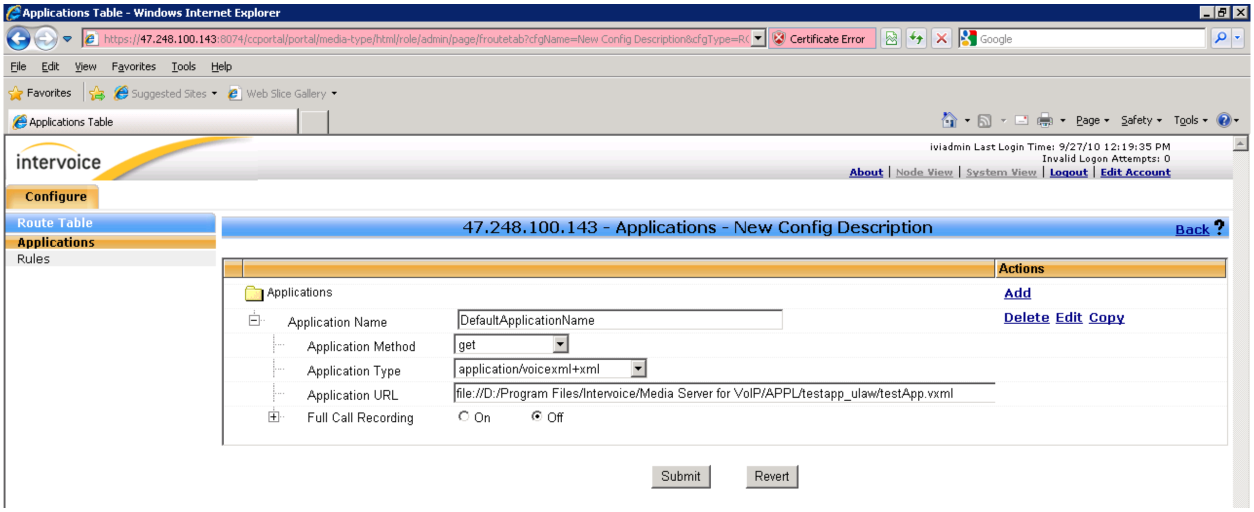

Figure 41: Applications Configuration Page

6. General Test Approach and Test Results

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

6.1. General Test Approach

The general test approach was to have one telephone stations of the CS 1000 to place a call to the Convergys MS and follow its voice instructions to verify other features of the Convergys MS. The main objectives were to verify the Convergys MS successfully performed the following:

- SIP Registration of the Convergys MS to the Avaya CS 1000 NRS SIP Proxy Server via SIP trunk.
- Basic SIP trunk calls
- DTMF 2833 and INBAND transmission methods.
- Transfer call to SIP and non-SIP telephones, ACD queue, and the emulated PSTN via PRI trunk.
- Telephony features: Call forward to voicemail with Message Waiting Indication (MWI) notification, busy, hold and retrieve.
- Codec negotiations G.711, G.729, and G723.
- Serviceability.

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 Convergys MS and Avaya CS 1000 system, please also refer to the Figure 1 for more detail.

- Step1: Registering the Convergys MS to the NRS SIP Proxy Server as SIP gateway endpoint.
- Step 2: Place a call from an IP phone of CS 1000 to the Convergys MS by entering the DN number.
- Step 3: A voice greeting and instruction from Convergys MS should be heard on the IP phone 2. The following options were presented in the script on the lab test bed. However, there were options 3, 5, 6, and 0 were programmed to use for this testing.
  - Press 1 to test speech recognition
  - Press 2 to test text to speech
  - Press 3 to test transfer
  - Press 4 to test CTI
  - Press 5 to test DTMF
  - Press 6 to test record function
  - Press 0 to exit
- Step 4: Select option 3 and then select option 1 to test a blind transfer, using the keypad on the IP phone to input a DN of a SIP phone.
- Step 5:
  - Answer the call on the SIP phone, the two way speech path established between the IP phone and SIP phone.
  - The SIP phone does not answer the call, after 4 rings the call is transferred to the DN of the CallPilot voice mail system on CS 1000. The IP phone can leave a message for the SIP phone and the MWI is lit on the SIP phone after the IP Phone hangs up.
- Step 6: On the IP phone, press the Hold button to hold the call and then press Hold button again to retrieve the call, the speech path is expected to be heard.

Repeat similar steps with a phone and combine with other options of Convergys IVR MS.

8. Conclusion

All of the executed test cases have passed and met the objectives in the section 6.1. The main features of Convergys MS were successfully tested with basic call features of 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 Convergys products can be found at www.Convergys.com

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.