Avaya Solution & Interoperability Test Lab

Application Notes for Skype Connect Service with Avaya™ Communication Server 1000 Release 6.0 – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Server 1000 Release 6.0 and Skype Connect Service. During the interoperability testing, Avaya™ Communication Server 1000 was able to interoperate with Skype Connect Service. This test was performed to verify the calls among CS1000, PSTN and Skype user such as basic call, call forward (all calls, busy, no answer), call transfer (blind and consult), conference, and voice mail. The call is placed in both directions with various set types.

These Application Notes have been obtained through Interoperability testing and additional technical discussions. Testing was conducted via the Interoperability Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

This document provides a typical network deployment of Communication Server 1000 (hereafter referred to as CS1000) and Skype Connect. During the interoperability testing, all CS1000 telephony features were tested such as basic calls, call forward, call transfer, conference, CLID displayed, abandoned call, and voicemail.

In this configuration, CS1000 is configured as a SIP gateway endpoint and registered to Skype's SBC. For each SIP Trunk to Skype, an additional signaling server will be required.

This document just provides a general guideline. Further information, may be obtained from your Avaya support representative.

1.1. Interoperability Testing

The focus of this testing is to verify that CS1000 Release 6.0 was able to interoperate with Skype Connect. The following interoperability areas were covered.

- General call processing between CS1000 and Skype systems including:
  - Codec/ptime negotiation and trans-coding (G.711 u-law and G.729 / 20ms)
  - Hold/Retrieve on both ends
  - CLID displayed
  - Ring-back tone
  - Speech path
  - Dialing plan support
  - Advanced features (Call on Mute, Call Park, Call Waiting, Feature Access Code)
  - Abandoned Call
- Call redirection verification: all supported methods (blind transfer, consultative transfer, call forward, and conference) including CLID. Call redirection is performed from both ends
- Fax is not supported, neither T.38 or G.711
- DTMF on both direction
- SIP Transport UDP
- Thru dialing via PBX Call Pilot
- Voice Mail Server (hosted on Avaya system)
- Early Media Transmission
- Inter-office tandem Call

1.2. Caveats

- Skype Connect is currently U.S. only. The service will be introduced in other regions at a later stage.

- Skype Connect does not support calls to the emergency service. Another PSTN trunk must be provisioned in Avaya Communication Server CS1000 to route calls to the emergency service.
- Access to a broadband internet connection is required.
- Maximum of 300 simultaneous calls per SIP Profile. A company may have multiple SIP Profiles.
- Maximum 99 Online Numbers per SIP Profile. Sequential number block (DID) purchases will be introduced at a later stage.
- Premium-rated numbers (1-900, 1-976) are blocked.
- DNS A records are supported for Skype Connect service node name resolution, while DNS SRV records will be introduced at a later stage.
- The SIP REFER request is not supported for call redirection/transfer.
- SIP 3xx Redirect Responses are not supported.
- SIP over TLS is not currently supported by Skype Connect
- SRTP is not supported.
- T.38 fax is not supported.
- RTCP and RTCP XR are not supported.
- IP TOS or DiffServ QoS markings are neither set nor honored, therefore Skype Connect cannot guarantee the end-to-end voice quality. Service Level Agreements (SLAs) are not available.
- G.711A/mu-law, G.729 codecs are supported. Skype Connect always offers G729 as first choice.
- For outbound calls (local, national and international) via Skype Connect the E.164 or International numbering format (00 + <country code>) must be used.
- For inbound calls Skype Connect delivers the called/calling number in E.164 format
- Skype Connect calls are limited to 4 hours.
- Skype Connect is not guaranteed to work with credit card machines, franking (stamping) machines and alarm systems or other services which use a regular phone line with a modem connection.
- Currently, this solution does not support outbound SIP calls to Skype names.
- Calls from Avaya CS1000 extensions that activate Calling Party Name Display (CPND) Restriction will result in a caller id of 000-012-3456 or another bogus number.
- Occasionally on calls from the PSTN to the Avaya CS1000, post dial delays bigger than 7 seconds were observed before a SIP INVITE message comes in from Skype Connect.
- Early Media is not supported by Skype. Skype does not send a 183 Session Progress. User won't hear audible remote ring back but the call still completes.
- Skype Connect does not support "True Inband DTMF"

1.3. Dependencies

- o CS1000 R6.0 software and implementation of latest patches – on June 15, 2010
- o Skype Connect provides support to setup, configure, and troubleshoot on carrier switch for the duration of the testing.

1.4. Support

For technical support on Skype Connect Service, please contact Skype technical support at:

- Toll Free: 1-866-214-7070 (CA)
- E-mail: **support@skype.net**

2. Reference Configuration

Figure 1 illustrates the test configuration used during the compliant testing event between the Communication Server 1000 and Skype Connect Service. This configuration is for a single Communication Server1000 deployment.

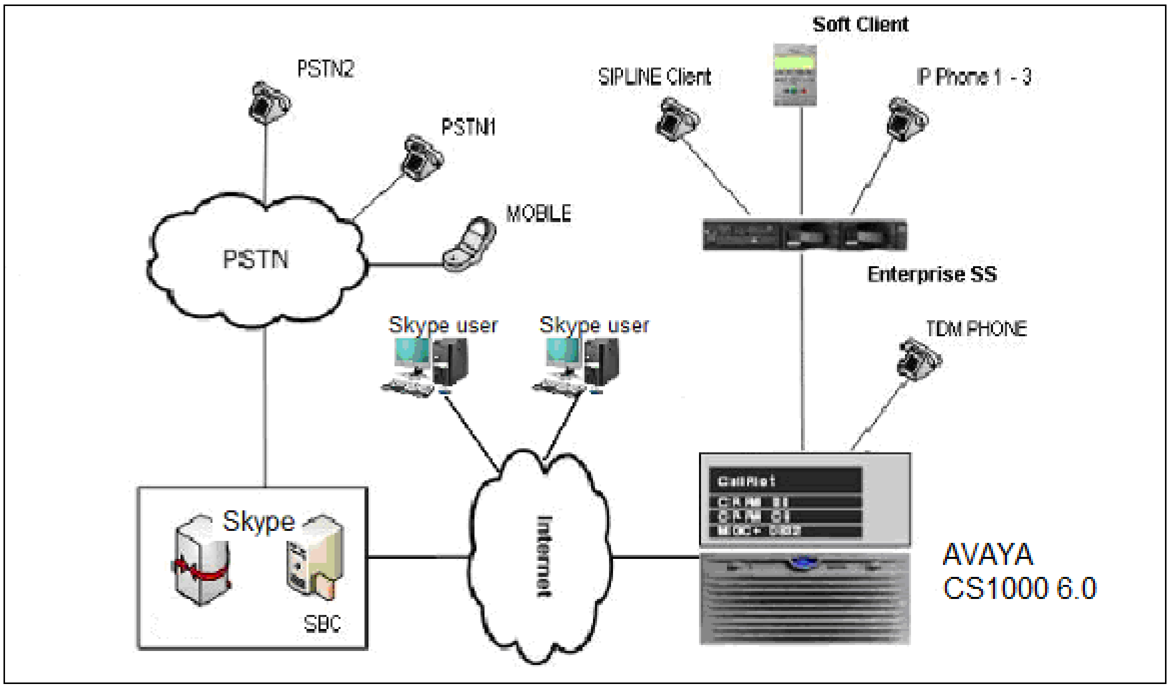

Figure 1- Network diagram for Avaya CS1000 – Skype Connect

Below is the deployment option for 2 or more CS1000 with Skype Connect.

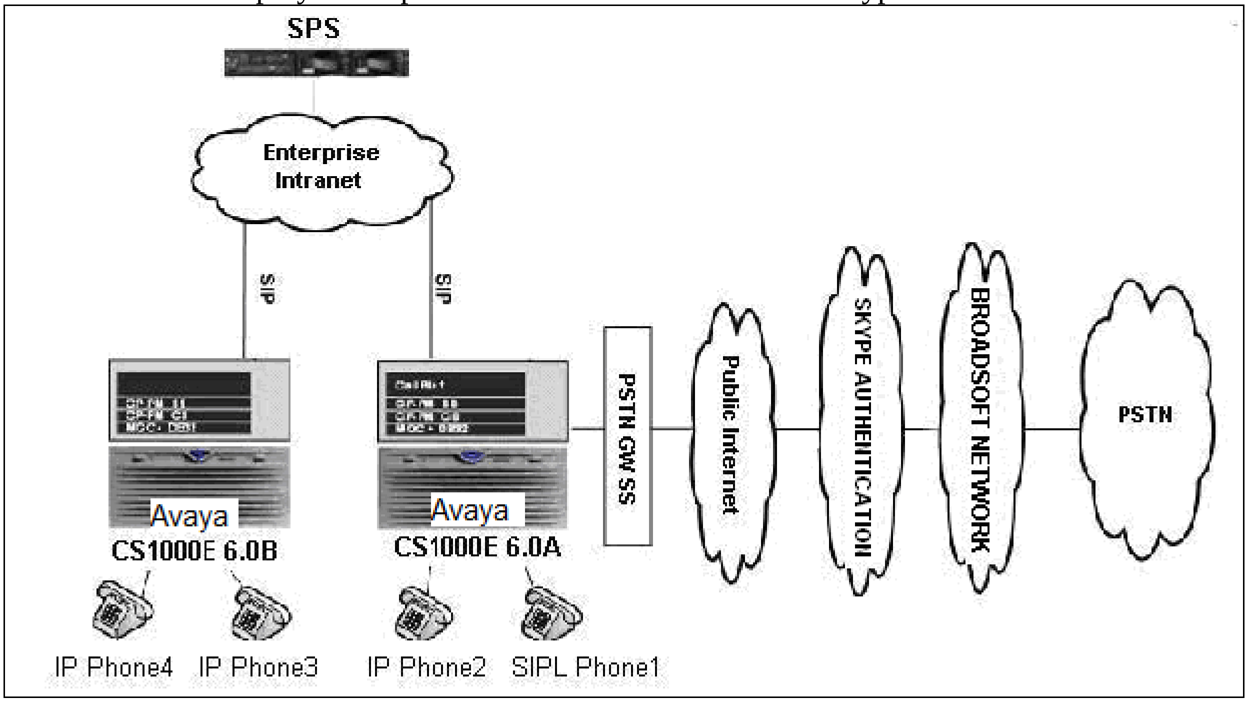

Figure 2 - Network topology for Multi-System configuration for Tandem Calls

The following assumptions were made for this lab test configuration.

1. CS1000 R6.0 software and implementation of latest patches
2. Skype Connect provides support to setup, configure, and troubleshoot on carrier switch for the duration of the testing.

All test scenarios involving the establishment of calls which were assume the following activities.

1. Calls were checked for the correct call progress tones and cadences.
2. During the ringing state the ring back tone and destination ringing were checked.
3. Calls were checked in both hands-free and handset mode due to internal Avaya requirement.
4. Calls were checked for speech path in both directions using spoken words to ensure clarity of speech.
5. The display(s) of the sets/clients involved were checked for consistent and expected CLID, name and redirection information both prior to answer and after call establishment.
6. The speech path and messaging system were observed for timely and quality End to End tone audio path generation and application responses.
7. The call server maintenance terminal window were open during the test cases execution for the monitoring of BUG(s), ERR and AUD messages.
8. Speech path and display checked before and after calls were put on/off hold from each end.
9. Applicable of files were screened on an hourly basis during the testing for message that may indicate technical issues. This refers to Avaya PBX files.
10. Calls were checked to ensure that all resources such as Virtual trunks, TDM trunks, Sets and VGWs are released when a call scenario ends

3. Equipment and Software Validated

Additional software and patch lineup for the configuration is as follows.

Call Server: CoRes 6.00R plus latest DEPLIST - June 15, 2010.
Signaling Server: SSE 6.00.18 plus latest DEPLIST - June 15, 2010
Patch lineup: the following patches are also loaded and applied.

Patch ID	Issue	Title
MPLR25946	1	SIP GW patch to remove outbound MCDN from SIP messaging
MPLR27159	1	Mandatory parameter "T38FaxRateManagement" isn't present in T38 SDP
MPLR30291	1	Support P-Asserted-ID and Caller ID Restriction

Hardware system requirement and software/load-ware version are as shown in the table below.

System	Software/Loadware version
Avaya CS1000 6.0 (CPPM)	• Call Server: 6.00R (CoRes) • Signaling Server: 6.00.18
Avaya phones	• 2002 p2: 0604DCJ (Unistim) • 2004 p2: 0604DCJ (Unistim) • 1140: 0625C6O (Unistim) • 1120: 0624C6O (Unistim) • 2007: 0621C6M (Unistim) • 1220: 062AC6O (Unistim) • SIP 1140 i00v142 • SIP 1120 • SMC3456: Version 2.6 - RC14 build 53715
Skype	• R1.4

Here is the output of "pstat" command on SSG.

```
[nortel@node1-carrier ~]$ pstat
Product Release: 6.00.18.00
In system patches: 6
PATCH#  NAME       IN_SERVICE   DATE      SPECINS  TYPE   RPM
19      p28774_1   Yes          10/08/10  NO       FRU    nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386
20      p28797_1   Yes          10/08/10  NO       FRU    nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386
21      p29703_1   Yes          10/08/10  NO       FRU    nortel-cs1000-shared-ssSubagent-6.00.18-00.i386
22      p25946_1   Yes          10/08/10  NO       FRU    nortel-cs1000-pi-control-1.00.00.00-00.noarch
24      p27159_1   Yes          10/08/10  NO       FRU    nortel-cs1000-pi-control-1.00.00.00-00.noarch
26      p30291_1   Yes          10/08/10  NO       FRU    nortel-cs1000-pi-control-1.00.00.00-00.noarch

In System service updates: 19
PATCH#  IN_SERVICE   DATE       SPECINS   REMOVABLE   NAME
0       Yes          10/08/10   yes       yes         nortel-cs1000-linuxbase-6.00.18.65-03.i386.001
1       Yes          10/08/10   NO        YES         submgr-2.00.02.00-01.i386.000
```

```
2       Yes         10/08/10   NO        YES         nortel-cs1000-gk-6.00.18.63-00.i386.000
3       Yes         10/08/10   NO        YES         nortel-cs1000-sps-6.00.18.63-00.i386.000
4       Yes         10/08/10   NO        YES         nortel-cs1000-shared-general-6.00.18.62-00.i386.000
5       Yes         10/08/10   NO        YES         nortel-cs1000-shared-pbx-6.00.18.62-00.i386.000
6       Yes         10/08/10   NO        YES         nortel-cs1000-emWeb_6-0-06.00.18.63-01.i386.001
7       Yes         10/08/10   NO        YES         nortel-cs1000-pd-6.00.18.62-00.i386.000
8       Yes         10/08/10   NO        YES         nortel-cs1000-dmWeb-6.00.18.62-00.i386.001
9       Yes         10/08/10   NO        YES         nortel-cs1000-csmWeb-6.00.18.62-00.i386.001
10      Yes         10/08/10   NO        YES         nortel-cs1000-auth-6.00.18.62-00.i386.000
11      Yes         10/08/10   NO        YES         nortel-cs1000-ISECSH-6.00.18.62-00.i386.000
12      Yes         10/08/10   NO        YES         nortel-cs1000-dbcom-6.00.18.65-01.i386.001
13      Yes         10/08/10   YES       YES         nortel-cs1000-tps-6.00.18.65-07.i386.000
14      Yes         10/08/10   YES       YES         nortel-cs1000-csv-6.00.18.65-04.i386.000
16      Yes         10/08/10   NO        YES         nortel-cs1000-bcc_6-0-6.00.18.65-02.i386.000
17      Yes         10/08/10   NO        YES         nortel-cs1000-cs1000WebService_6-0-6.00.18.65-
        02.i386.000
18      Yes         10/08/10   NO        YES         nortel-cs1000-ftrpkg-6.00.18.65-02.i386.000
25      Yes         10/08/10   NO        YES         nortel-cs1000-vtrk-6.00.18.65-TMP297.i386.001
[nortel@node1-carrier ~]$
```

4. Avaya Communication Server 1000 Configuration

These Application Notes used the Coordinated Dial Plan (CDP) feature to receive the calls and used the Special Number Translation (SPN) feature to route calls from the Avaya Communication Server 1000, over Skype SIP trunk to PSTN.

These Application Notes assume that the basic configuration has already been administered. For further information on Avaya Communications Server 1000, please consult references in Section 9.

The below procedures describe the configuration details of CS1000 with a SIP trunk to Skype.

- **4.1. Log in to CS1000 System**
  - 4.1.1. UCM and EM
  - 4.1.2. Call Server Overlay
- **4.2. Administer A Node IP Telephony**
  - 4.2.1. Obtain Node IP address
  - 4.2.2. Administer TPS
  - 4.2.3. Administer Quality of Service (QoS)
  - 4.2.4. Synchronize The New Configuration

- **4.3. Administer Voice Codec**
  - 4.3.1. Enable Voice Codec G711, G729 on Node IP Telephony
  - 4.3.2. Enable Voice Codec G711, G729 on Media Voice Gateways - MGC

- **4.4. Zones and Bandwidth Management**
  - 4.4.1. Create a zone for IP phones (zone 10)
  - 4.4.2. Create a zone for virtual SIP trunk (zone 255)

- **4.5. Administer SIP Trunk Gateway**
  - 4.5.1. Integrated Services Digital Network (ISDN)
  - 4.5.2. Administer SIP trunk gateway to Skype
  - 4.5.3. Administer Virtual D-Channel
  - 4.5.4. Administer Virtual Super-Loop (SUPL V100)
  - 4.5.5. Administer Virtual SIP Routes (ROUT 100)
  - 4.5.6. Administer Virtual Trunks

- **4.6. Administer Dialing Plans**
  - 4.6.1. Digit Manipulation Block **(**DMI) for Inbound Call (DMI 7)
  - 4.6.2. Digit Manipulation Block **(**DMI) for Outbound Call (DMI 25)
  - 4.6.3. Route List Block (RLB) for National Call (RLB 25)
  - 4.6.4. Route List Block (RLB) for International Call (RLB 26)
  - 4.6.5. Inbound Call

4.1. Log in to CS1000 System

Log in Unified Communications Management (UCM) and Element Manager (EM)

a) Open an instance of a web browser and connect to the UCM GUI at the following address: http://<node IP address> or http://<UCM IP address>. Log in using an appropriate Username and Password.

Figure 3 – Log in Unified Communications Management.

b) The **Unified Communications Management** screen is displayed. Click on the element Name of the CS1000 Element.

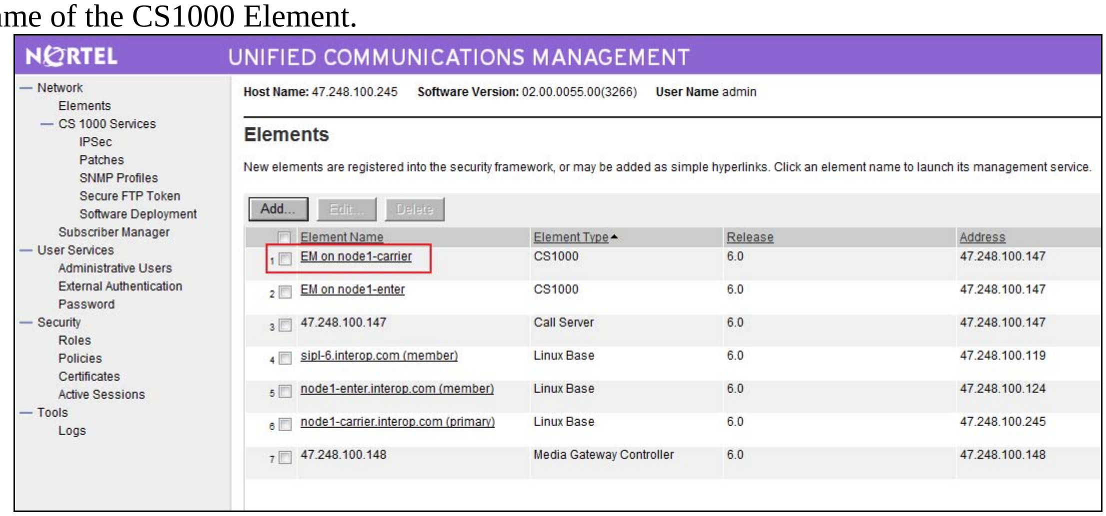

Figure 4 – Unified Communications Management Page.

c) The CS 1000 Element Manager **System Overview** page is displayed.

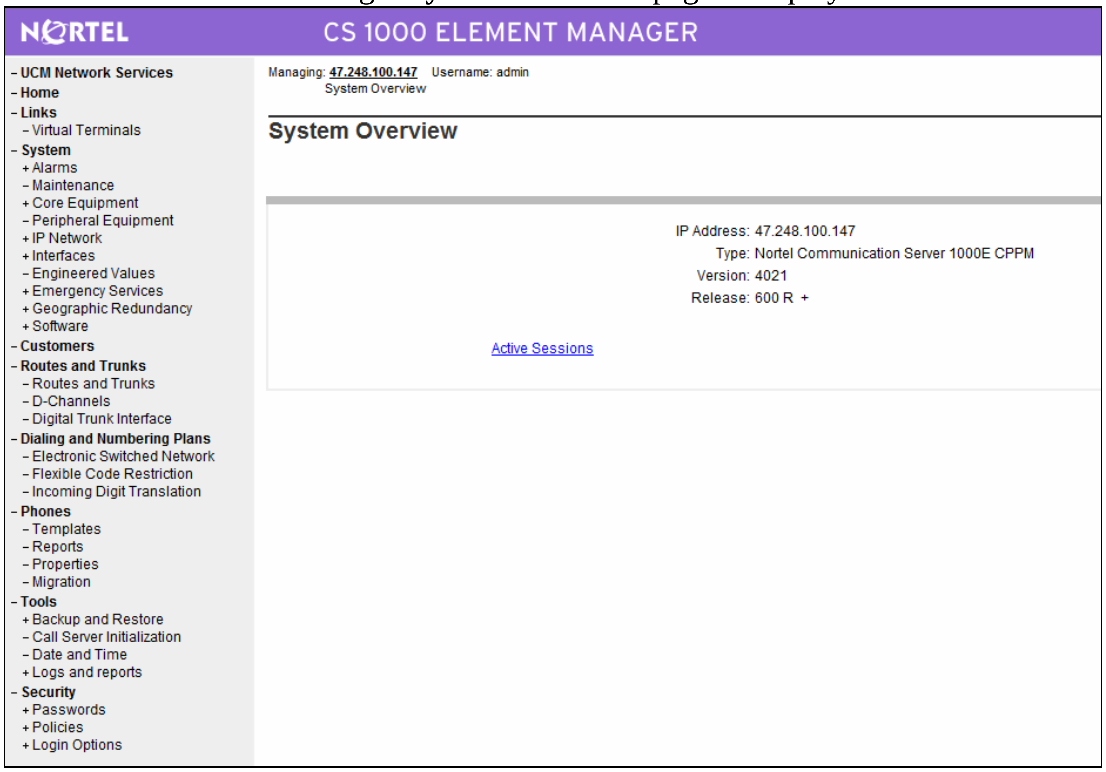

Figure 5 – Element Manager System Overview Page.

Call Server Overlay

a) SSH to IP address of SSG or Signaling Server with the nortel/admin account.
b) Run the command "cslogin" and log in with the admin account.
c) Here are the logs.

login as: **nortel**

Nortel Networks Linux Base 6.00

The software and data stored on this system are the property of, or licensed to, Nortel Networks and are lawfully available only to authorized users for approved purposes. Unauthorized access to any software or data on this system is strictly prohibited and punishable under appropriate laws. If you are not an authorized user then do not try to log in. This system may be monitored for operational purposes at any time.

nortel@47.248.100.245's password: **<----enter your password**
Last login: Tue Aug 31 15:33:59 2010 from 47.248.100.48

[nortel@node1-carrier ~]$ **cslogin**

SEC054 A device has connected to, or disconnected from, a pseudo tty without authenticating

TTY 08 SCH MTC BUG   10:40
OVL111 IDLE   0
>loii **admin**
PASS? <----enter your password
The software and data stored on this system are the property of,
or licensed to, Nortel Networks and are lawfully available only
to authorized users for approved purposes. Unauthorized access
to any software or data on this system is strictly prohibited and
punishable under appropriate laws. If you are not an authorized
user then log out immediately. This system may be monitored for
operational purposes at any time.

.
TTY #08 LOGGED IN ADMIN 10:40   2/9/2010

>

4.2. Administer A Node IP Telephony

4.2.1. Obtain Node IP address

These Application Notes assume that the basic configuration has already been administered. A Node has already been created. This Section describes the steps for configuring a Node (Node ID 1000) in CS1000 IP network to work with Skype. For further information on Avaya Communications Server 1000, please consult reference in Section 9.

a) Log in UCM and EM (please refer to Section 4.1.1)

b) Select **System** -> **IP Network** -> **Nodes: Servers, Media Cards** and then click on the Node ID of your CS1000 Element.

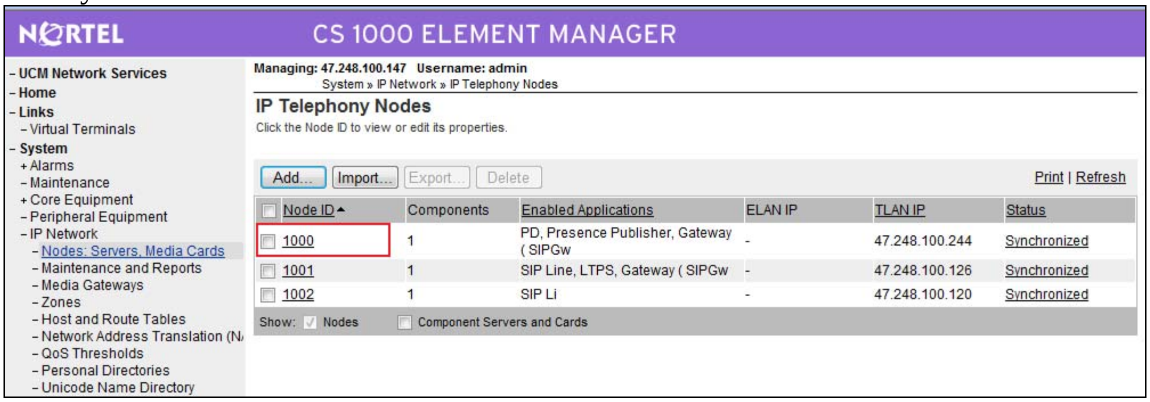

Figure 6 – IP Telephony Nodes Page.

c) The **Node Details** screen is displayed with the IP address of the CS1000 node. The **Node IP Address** is a virtual address which corresponds to the TLAN IP address of the Signaling Server, SIP Signaling Gateway. The SIP Signaling Gateway uses this **Node IP Address** to register with Skype.

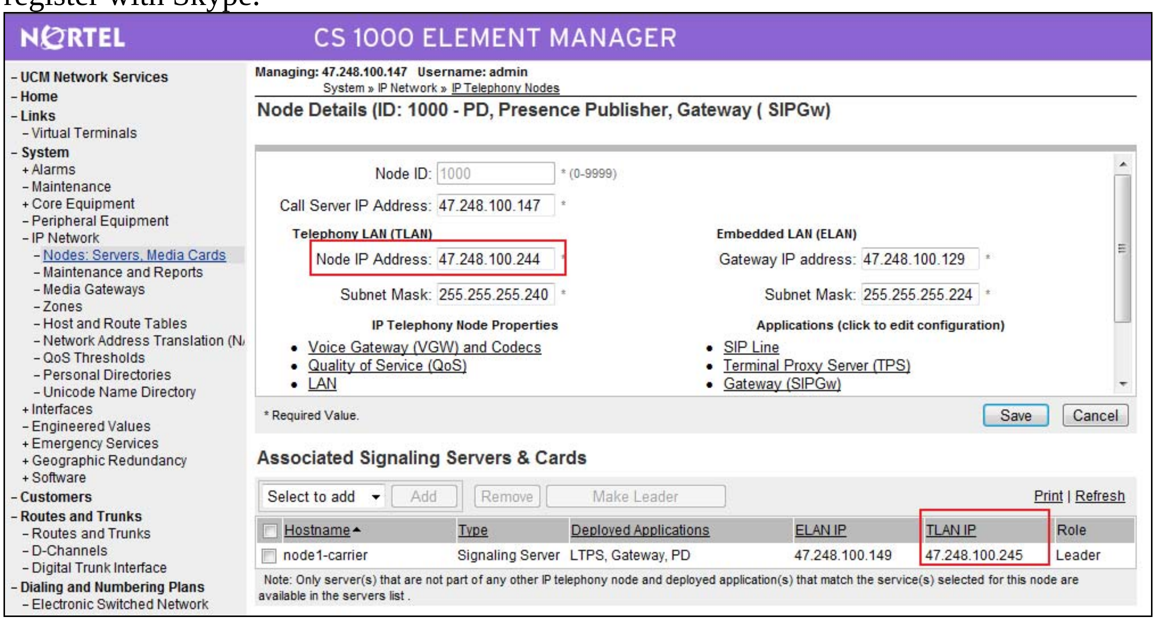

Figure 7 –Node Details Page.

4.2.2. Administer TPS

d) Continue Section 4.2.1, on the **Node Details** page, select **Terminal Proxy Server** (**TPS**)

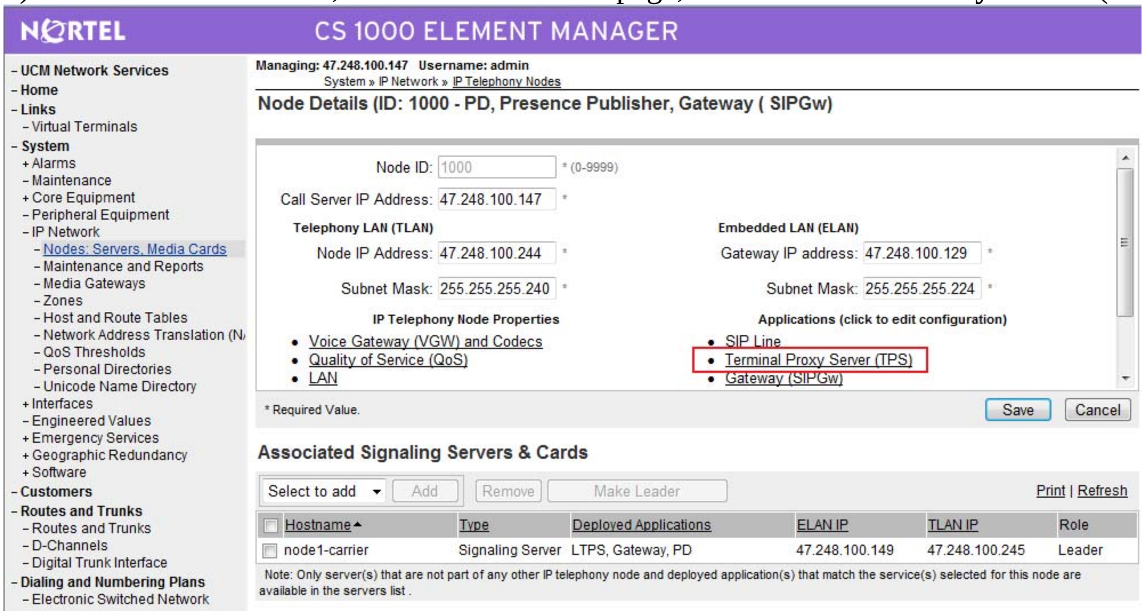

Figure 8 –Node Details Page – TPS.

e) Check the **UNIStim Line Terminal Proxy Server** and then click **Save.**

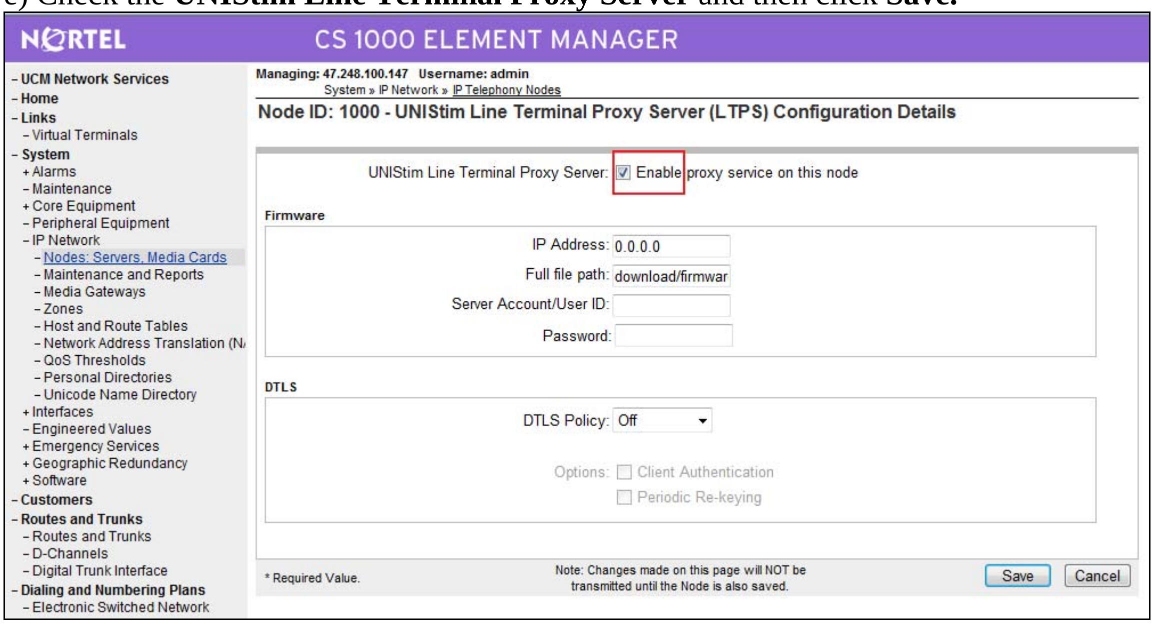

Figure 9 – TPS Configuration Details.

4.2.3. Administer Quality of Service (QoS)

f) Continue Section 4.2.2, on the **Node Details** page, click on **Quality of Service (QoS).**

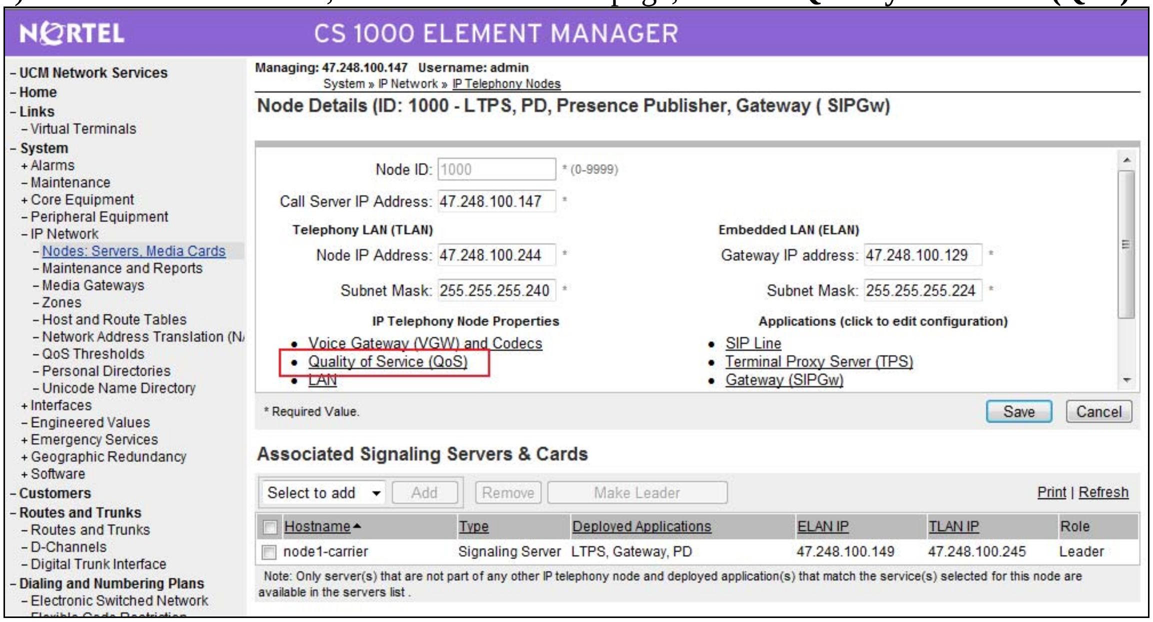

Figure 10 –Node Details Page – QoS.

g) The default Diffserv values are correct as shown in **Figure 11**. Click **Save.**

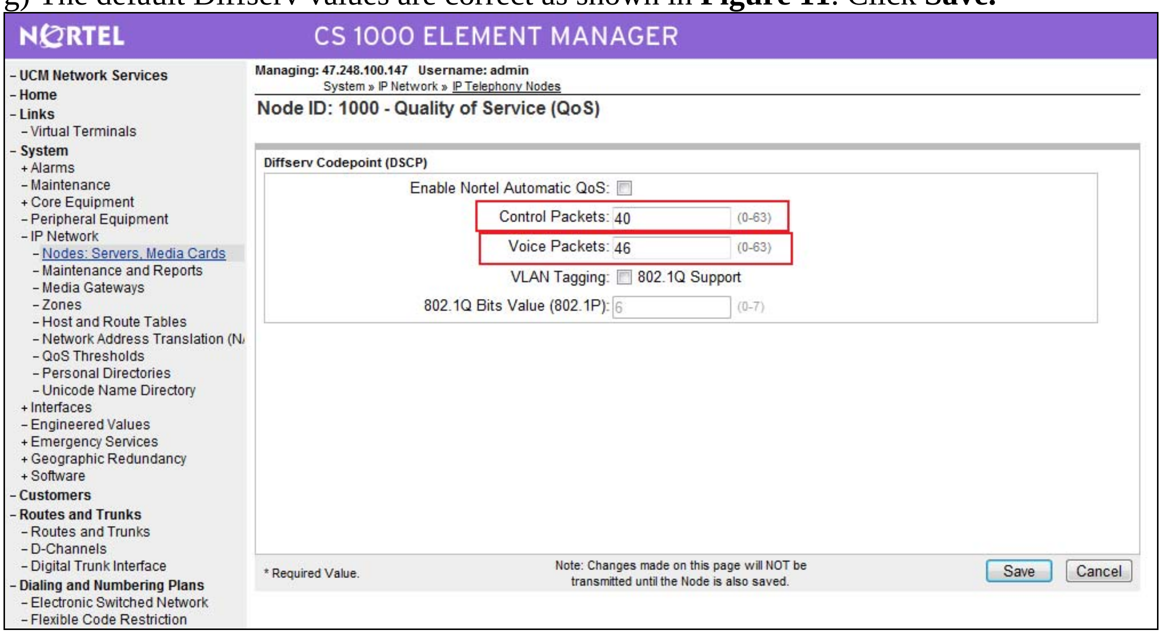

Figure 11 – QoS Configuration Details.

4.2.4. Synchronize The New Configuration

h) Continue Section 4.2.3, return to the **Node Details** screen and click **Save**.

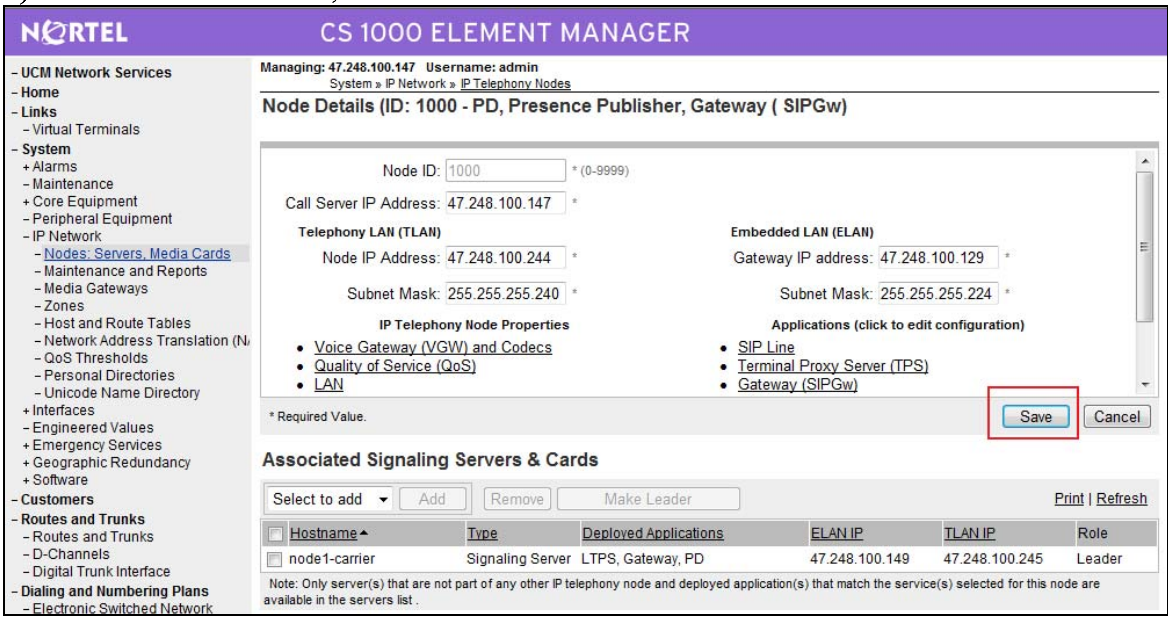

Figure 12 – Synchronize the new Configuration – Save.

i) The **Node Saved** screen is displayed. Click **Transfer Now**…

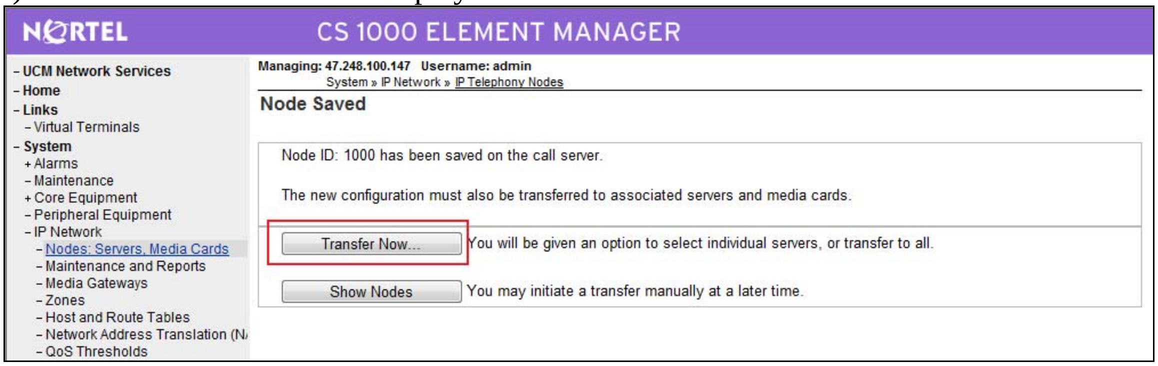

Figure 13 – Synchronize the new Configuration – Transfer.

j) The **Synchronize Configuration Files** screen is displayed. Select the Signaling Server and click on **Start Sync**.

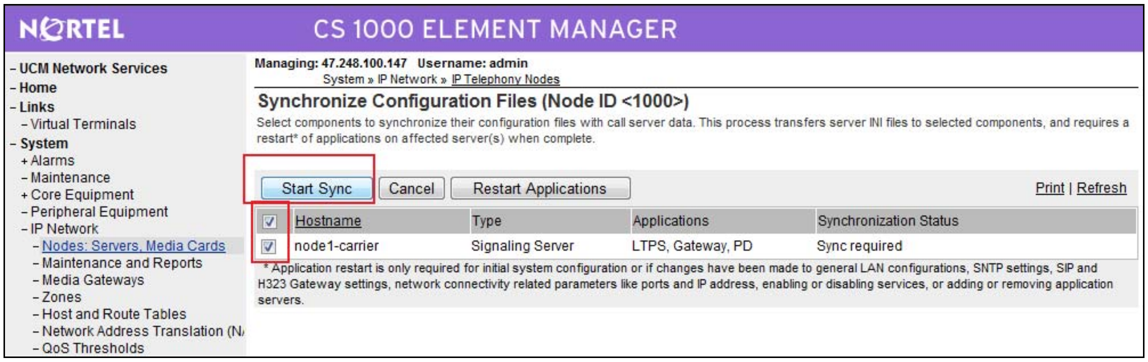

Figure 14 – Synchronize the new Configuration – Start Sync.

k) When the synchronization completes, Select the Signaling Server and click on **Restart Applications**.

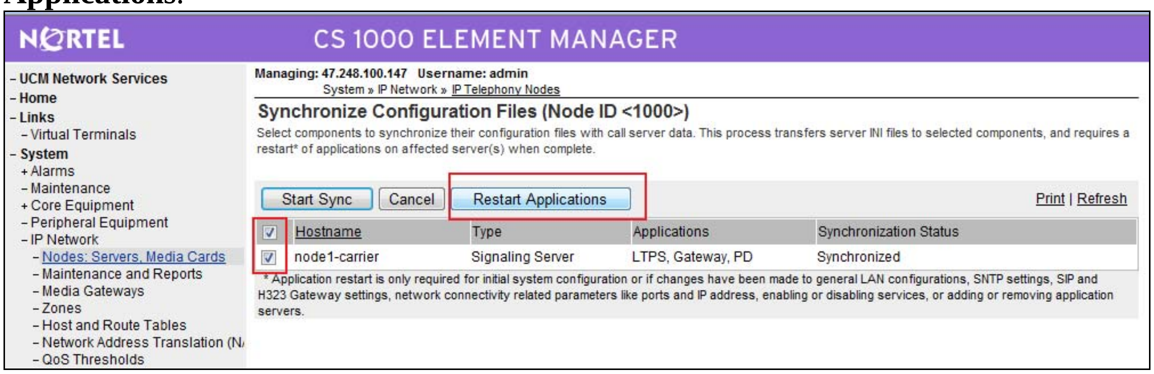

Figure 15 – Synchronize the new Configuration – Restart Applications.

4.3. Administer Voice Codec

4.3.1. Enable Voice Codec G711, G729 on Node IP Telephony.

a) Log in UCM and EM (Please refer to Section 4.1.1)

b) Select **IP Network** -> **Nodes: Servers, Media Cards** -> Configuration from the left pane, and in the **IP Telephony Nodes** screen displayed, select the **Node ID** of this CS1000 system. The **Node Details** screen is displayed. (See in Section 4.2.1 for more detail).

c) On the **Node Details** screen, click on **Voice Gateway (VGW) and Codec.**

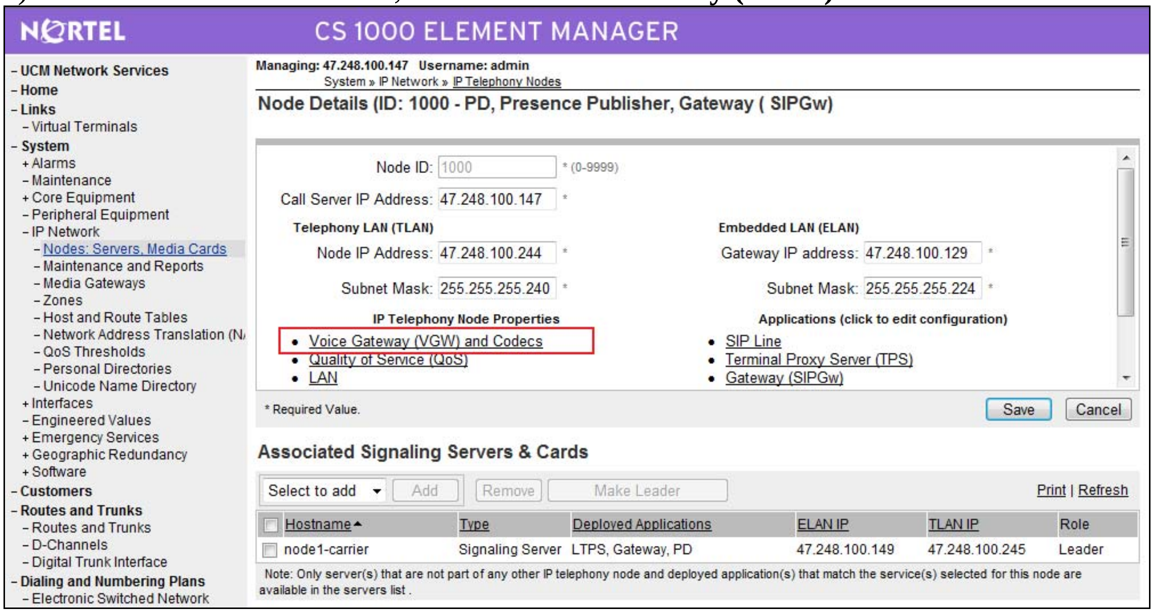

Figure 16 – Node Details - Voice Gateway and Codec.

d) In the following screen scroll down the parameters box and check the desired codecs under Voice Codecs. Note that G.729 and VAD are checked. Click on **Save.**

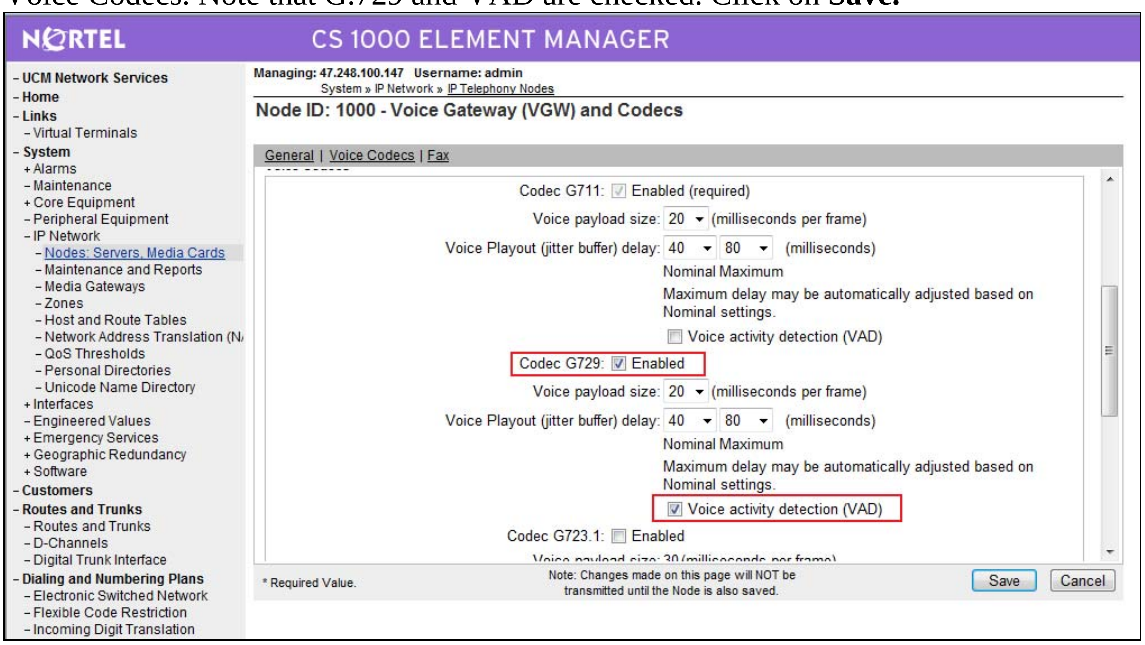

Figure 17 –Voice Gateway and Codec Configuration Details.

e) Synchronize the new configuration (please refer to Section 4.2.4)

4.3.2. Enable Voice Codec G711, G729 on Media Gateways.

a) Log in UCM and EM (please refer to Section 4.1.1 for more detail)
b) Select **IP Network** -> **Media Gateways** configuration from the left pane , click **MGC** from the right pane.

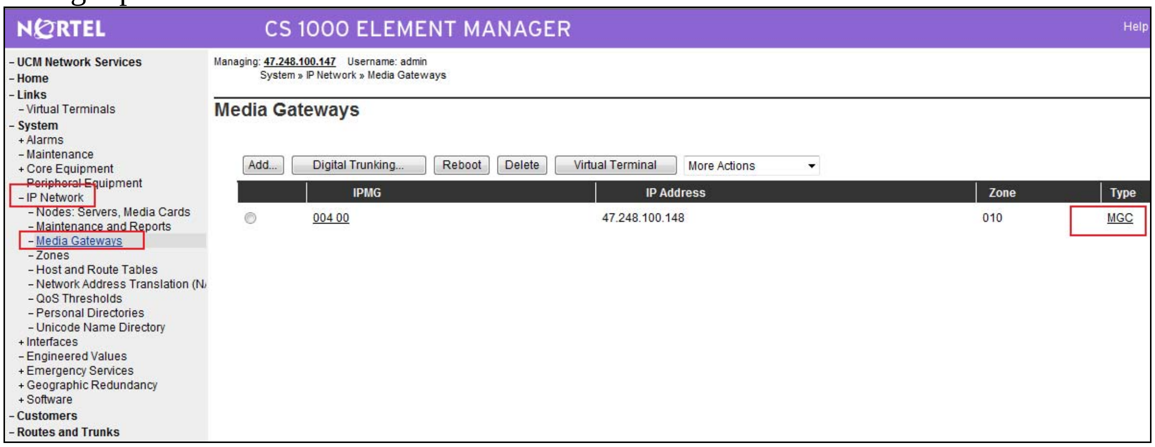

Figure 18 – Media Gateways Configuration Page.

c) In the following screen scroll down the parameters box and check **Codec G729A** and **VAD**

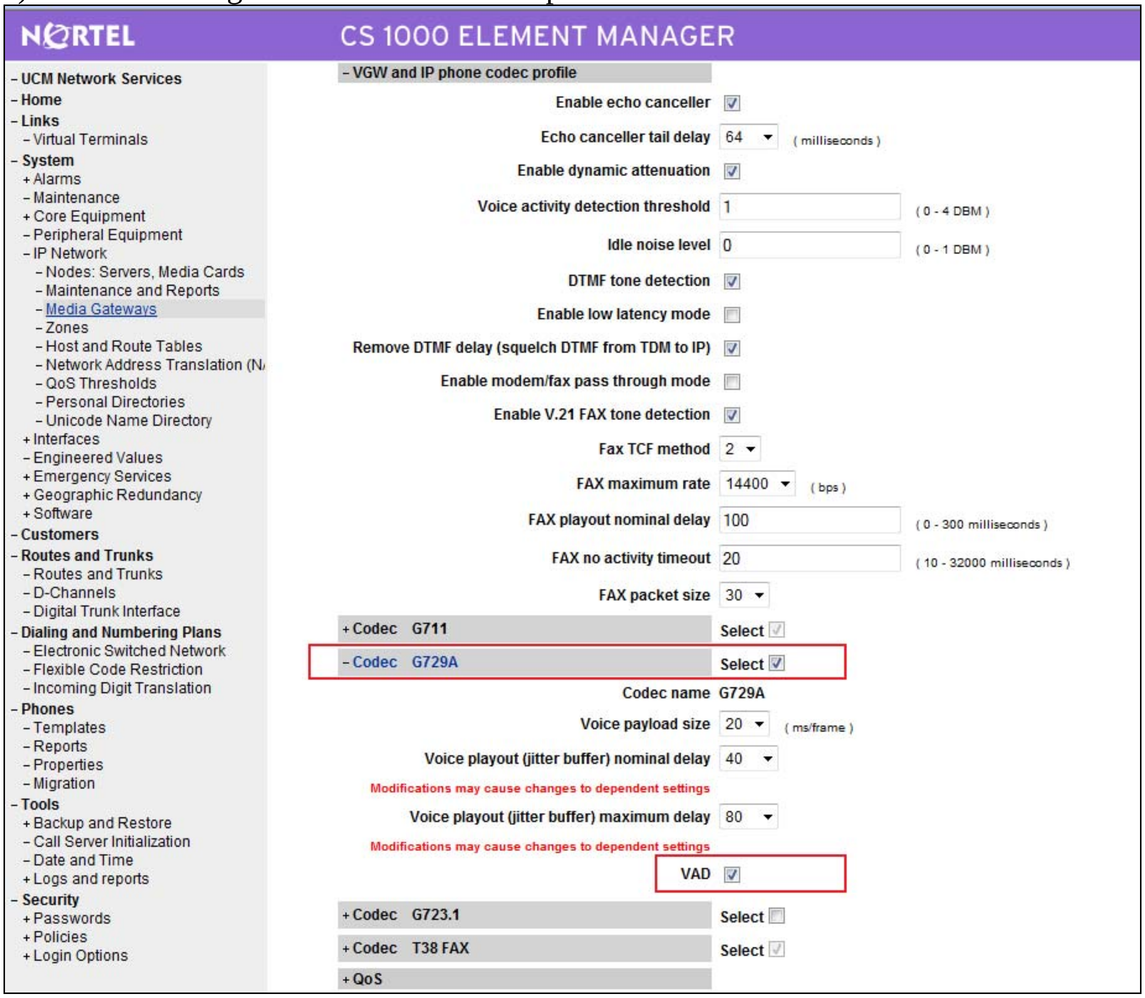

Figure 19 – Media Gateways Configuration Details Page.

d) In the following screen scroll down the parameters box and click on **Save**.

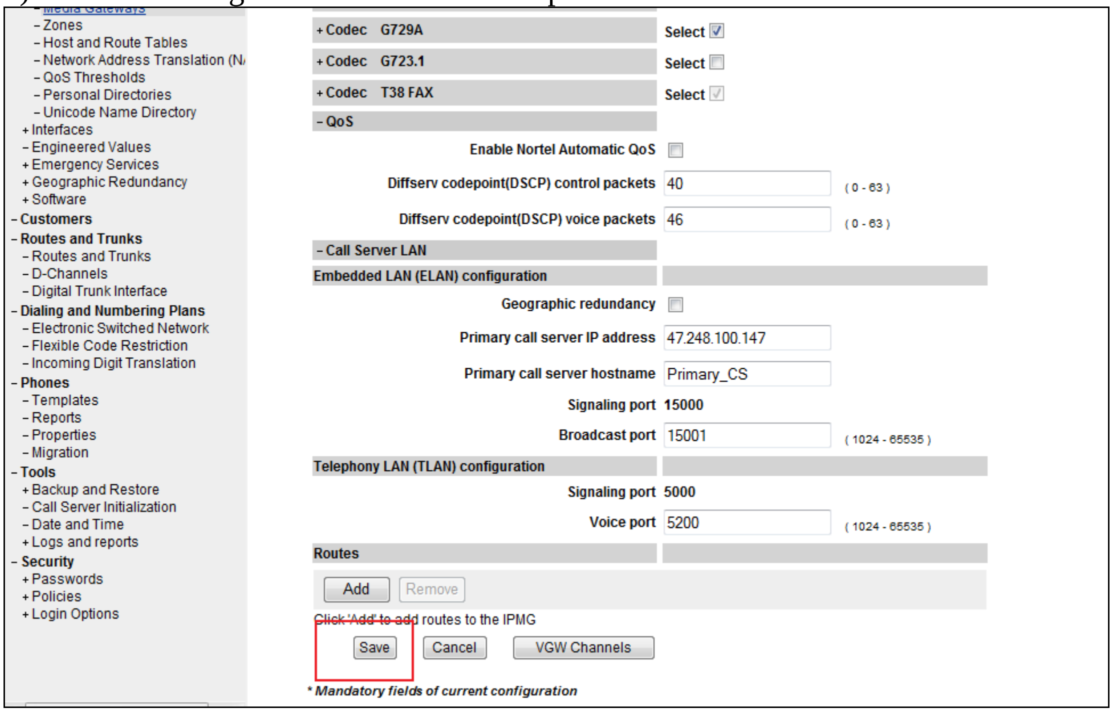

Figure 20 – Media Gateways Configuration Details Page – Save.

4.4. Zones and Bandwidth Management

This section describes the steps to create 2 zones: one for IP sets and anther one for SIP Trunk.

4.4.1. Create a zone for IP phones (zone 10)

The following figures show how to configure a zone for IP sets and bandwidth management. If it does not already exist, please click “**to Add**” button to create a zone for IP sets. The bandwidth strategy can be adjusted to preference.

a) Log in UCM and EM (please refer to Section 4.1.1 for more detail)
b) Select **IP Network** -> **Zones** configuration from the left pane, click **Bandwidth Zones**

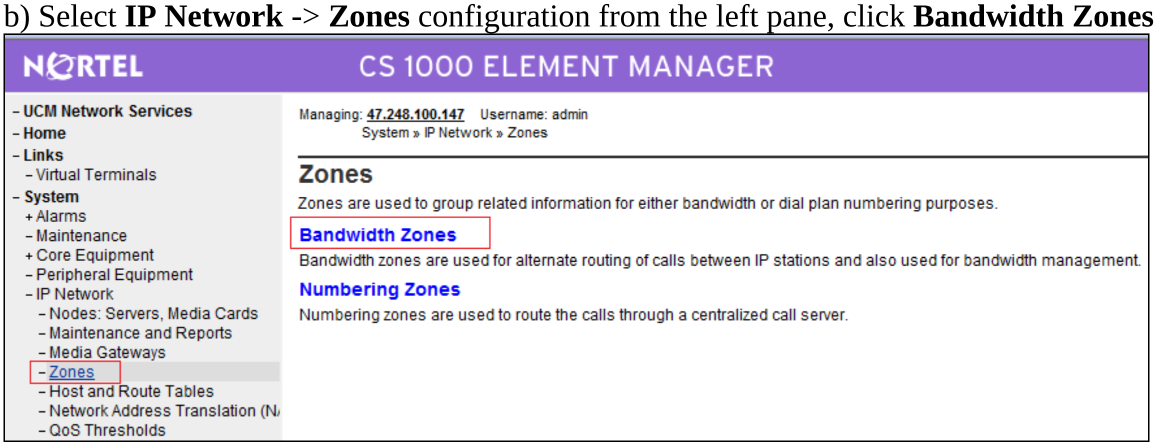

Figure 21 –Zones Page.

c) The **Bandwidth Zones** screen is displayed. Select "**Zone Basic Property and Bandwidth Management**".

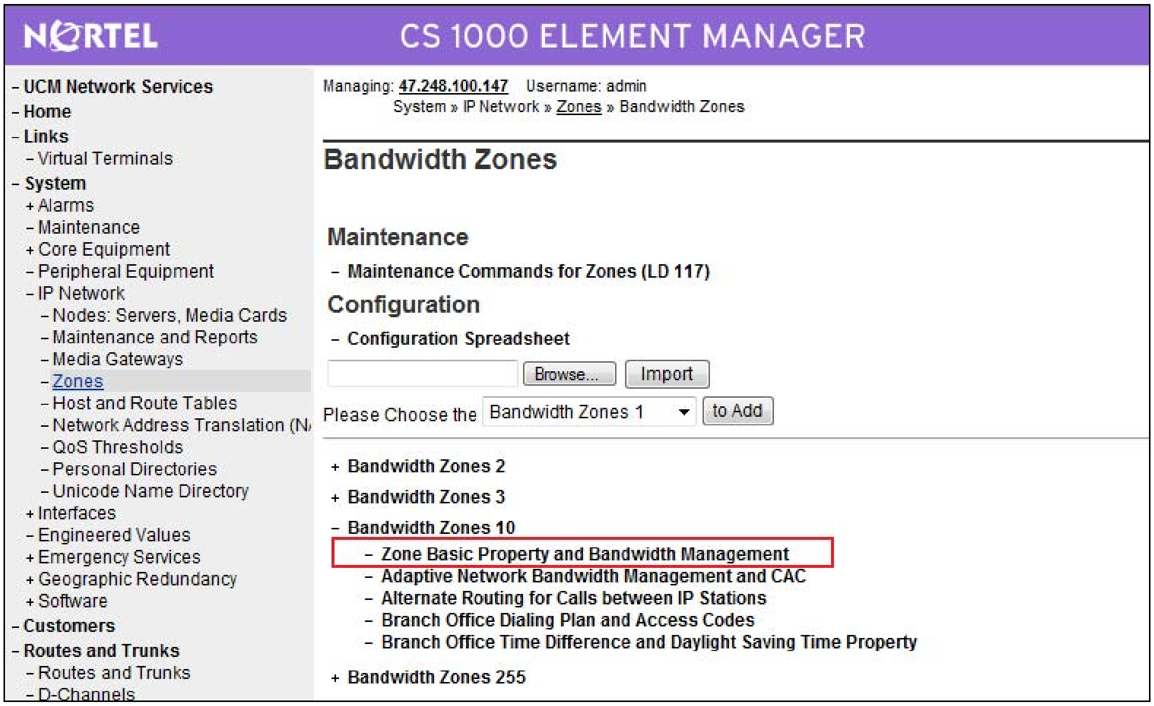

Figure 22 –Bandwidth Zones Page.

d) Select **MO** for **Zone Intent (ZBRN)** and click **Submit**

- INTRA_STGY: Codec configuration for local calls.
- INTER_STGY: Codec configuration for the calls over trunk.
- BQ: G711 is first choice and G729 is second choice.
- BB: G729 is first choice and G711 is second choice.
- MO: is used for IP phones, VGW ....etc
- VTRK: is used for virtual trunk.

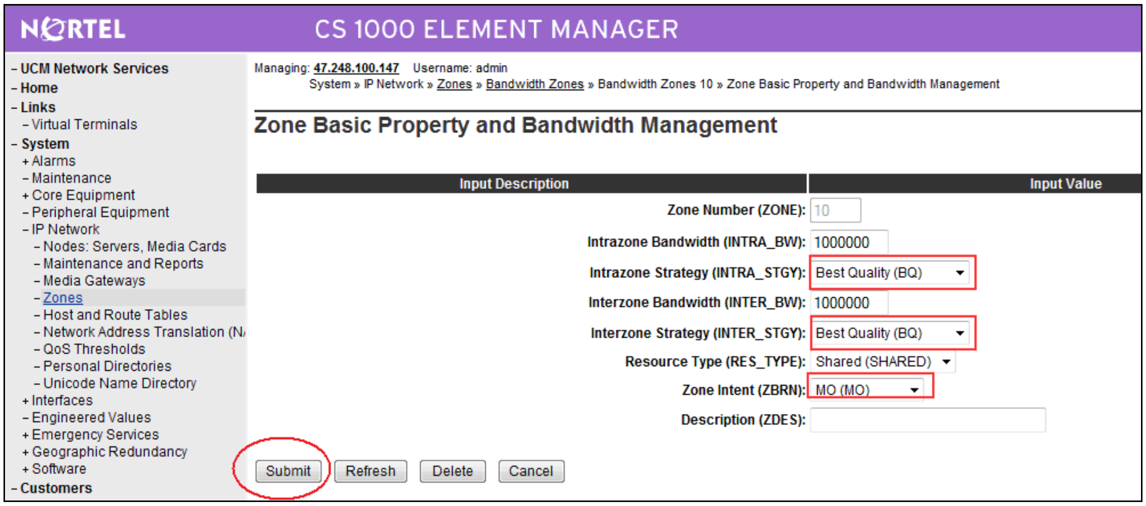

Figure 23 –Bandwidth Management Configuration Details Page – IP phone.

4.4.2. Create a zone for virtual SIP trunk (zone 255)

Follow Section 4.4.1 to create a zone for virtual trunk. The difference is in **Zone Intent (ZBRN)** field. Select **VTRK** for virtual trunk and then click **Submit**.

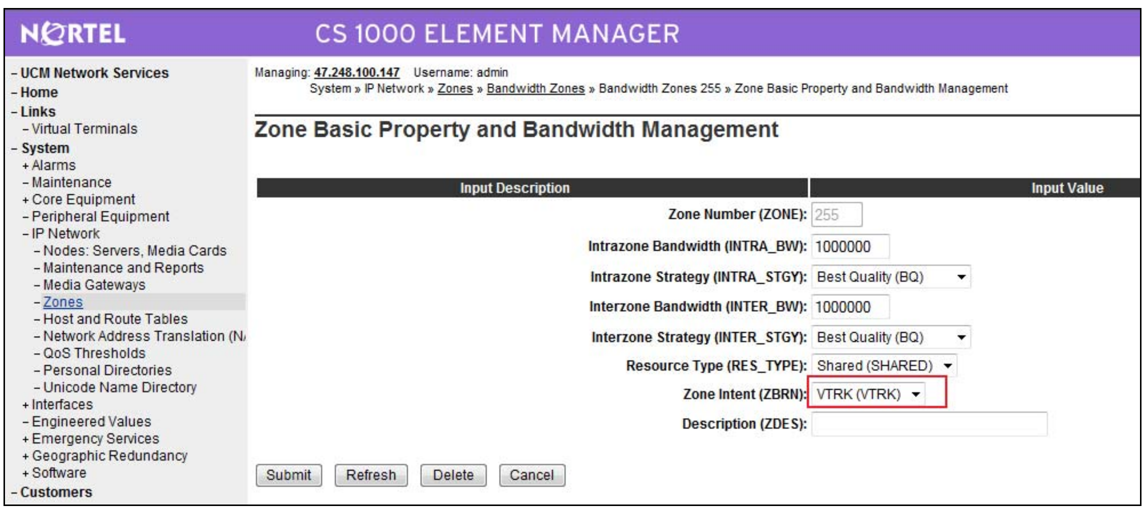

Figure 24 –Bandwidth Management Configuration Details Page –virtual trunk.

4.5. Administer SIP Trunk Gateway

This section describes the steps for establishing a SIP connection between CS1000 and Skype.

4.5.1. Integrated Services Digital Network (ISDN)

a) Log in UCM and EM (please refer to Section 4.1.1 for more detail)

b) Select **Customers** in the left pane. The **Customers** screen is displayed. Click the link associated with the appropriate customer, in this case **00**. The system can support more than one customer with different network settings and options.

The **Customer 00** screen is displayed next. Select **Feature Packages**.

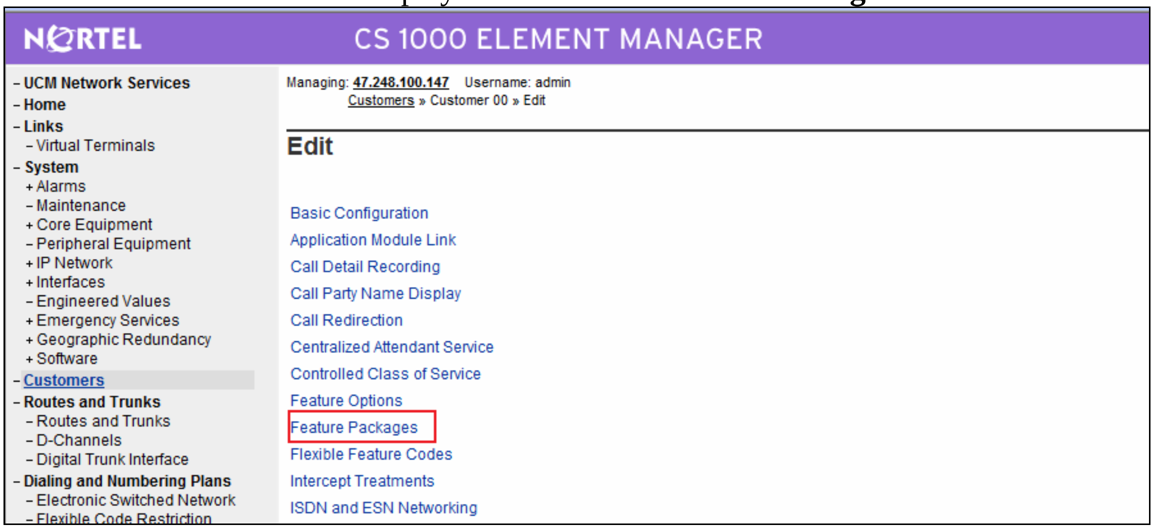

Figure 25 –Customer - feature packages Configuration Page.

c) The screen is updated with a listing of feature packages populated below **Feature Packages** (not all features shown below). Select **Integrated Services Digital Network** to edit its parameters. The screen is updated with parameters populated below **Integrated Services Digital Network**. Check the **Integrated Services Digital Network** (ISDN) checkbox, and retain the default values for all remaining fields. Scroll down to the bottom of the screen, and click **Save** (not shown).

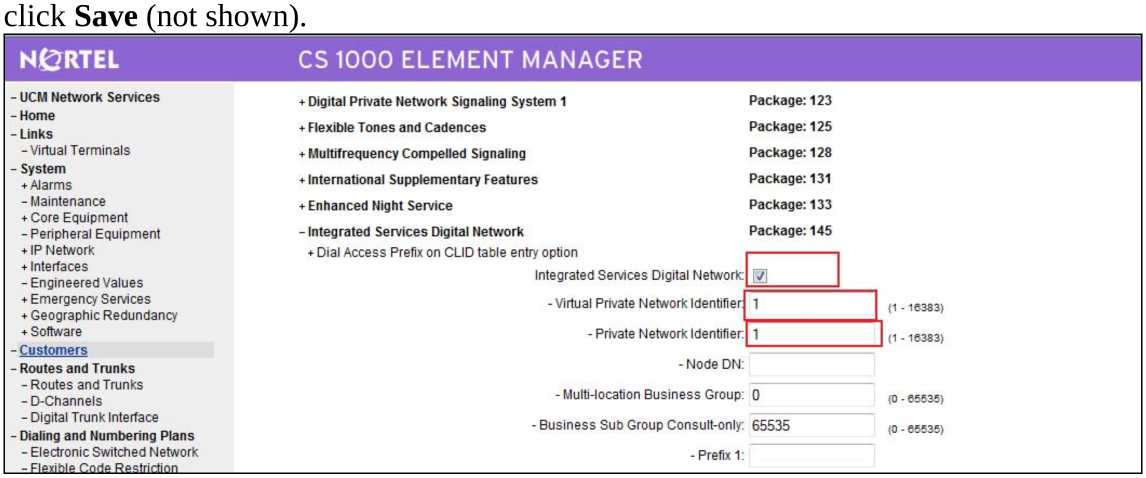

Figure 26 –Customer – ISDN Configuration Page.

4.5.2. Administer SIP Trunk Gateway to Skype

a) Log in UCM and EM (please refer to Section 4.1.1 for more detail)
b) Select **IP Network** -> **Nodes: Servers, Media Cards** configuration from the left pane, and in the **IP Telephony Nodes** screen displayed, select the **Node ID** of this CS1000 system. The **Node Details** screen is displayed. (Please refer to Section 4.2.1, **Figure 6**).
c) On the **Node Details** screen, select **Gateway (SIPGw)**

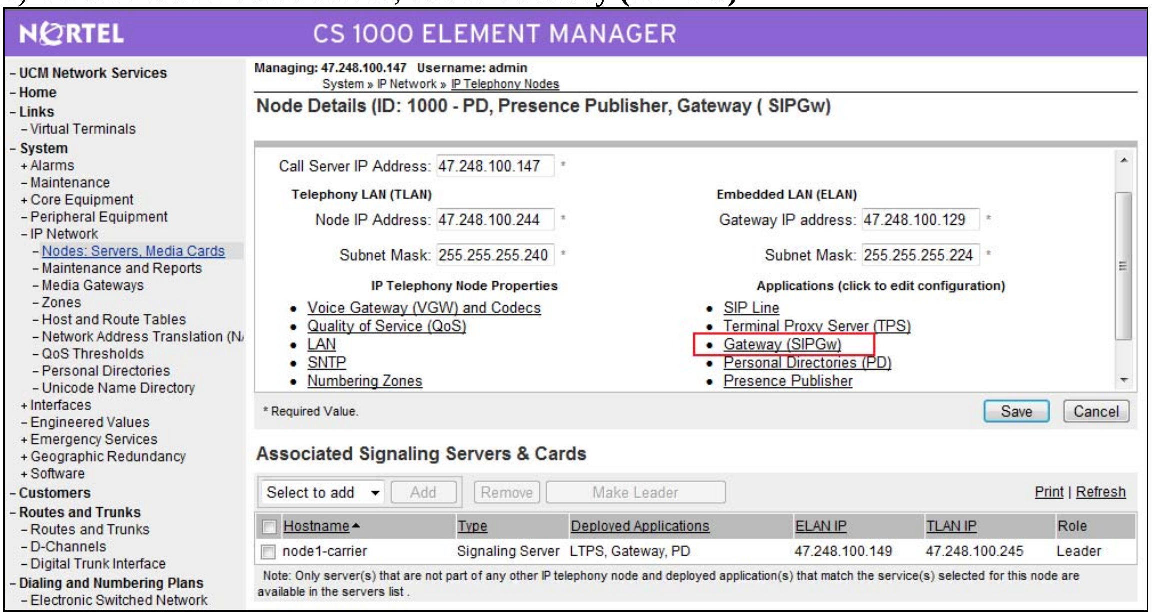

Figure 27 – Node Details – Gateway Configuration Page.

d) Under **General** tab of the **Virtual Trunk Gateway Configuration Details** screen, enter the following values for the specified fields, and retain the default values for the remaining fields.
- **Vtrk Gateway Application**: Select **SIPGw**
- **SIP Domain Name**: provided when user creates a SIP profile on Skype.
- **Local SIP Port**: provided when user creates a SIP profile on Skype.
- **Gateway endpoint name**: provided when user creates a SIP profile on Skype.
- **Gateway password:** provided when user creates a SIP profile on Skype.

The following parameters will be provided when user creates a SIP profile on Skype.

- Skype for SIP domain: sip.skype.com ---> SIP Domain Name
- Local SIP Port: 5060
- SIP User: 99051000106920 ---> Gateway Endpoint Name
- Password: xxxxxxxxxxxxxxx ---> Gateway password
- Primary Skype for SIP IP: 204.9.161.164 ---> Proxy Primary TLAN IP Address
- Secondary Skype for SIP IP: 63.209.144.201 ---> Proxy Secondary TLAN IP Address

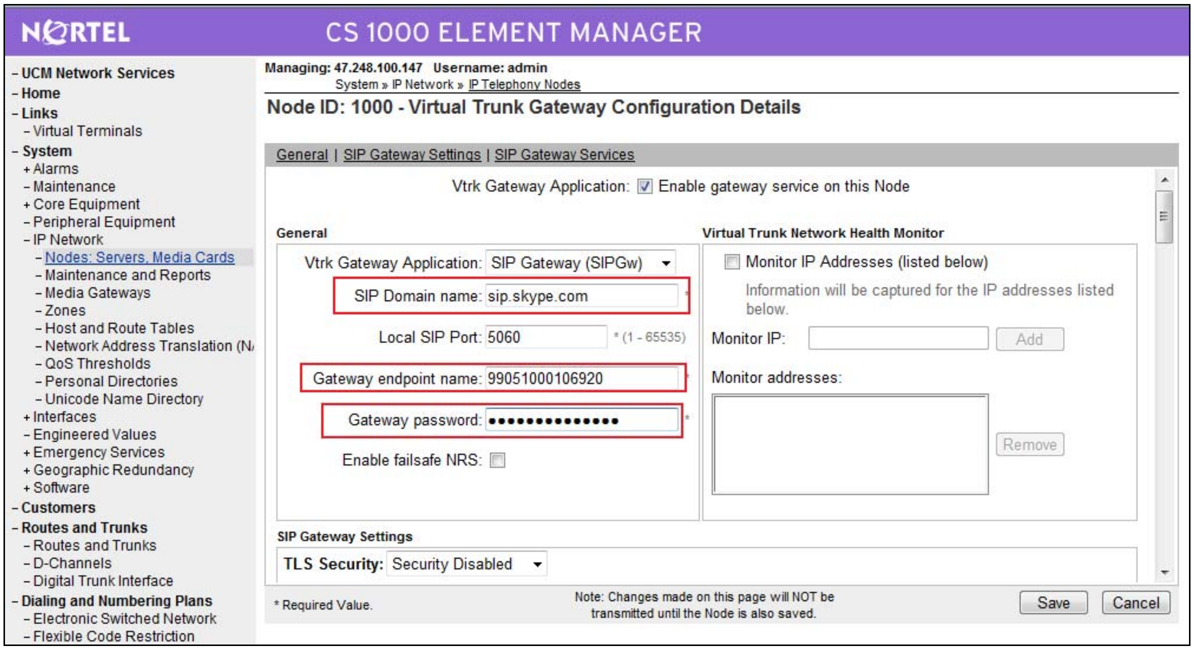

Figure 28 – Virtual Trunk Gateway Configuration Details Page.

e) Click on **SIP Gateway Settings** tab, under **Proxy or Redirect Server**, enter the following values for the specified fields, and retain the default values for the remaining fields.

Primary TLAN IP Address: provided when user creates a SIP profile on Skype.
Secondary TLAN IP Address: provided when user creates a SIP profile on Skype.
Port: **5060**
Transport Protocol: **UDP**
Options: Check **Support registration** and **Primary CDS Proxy**

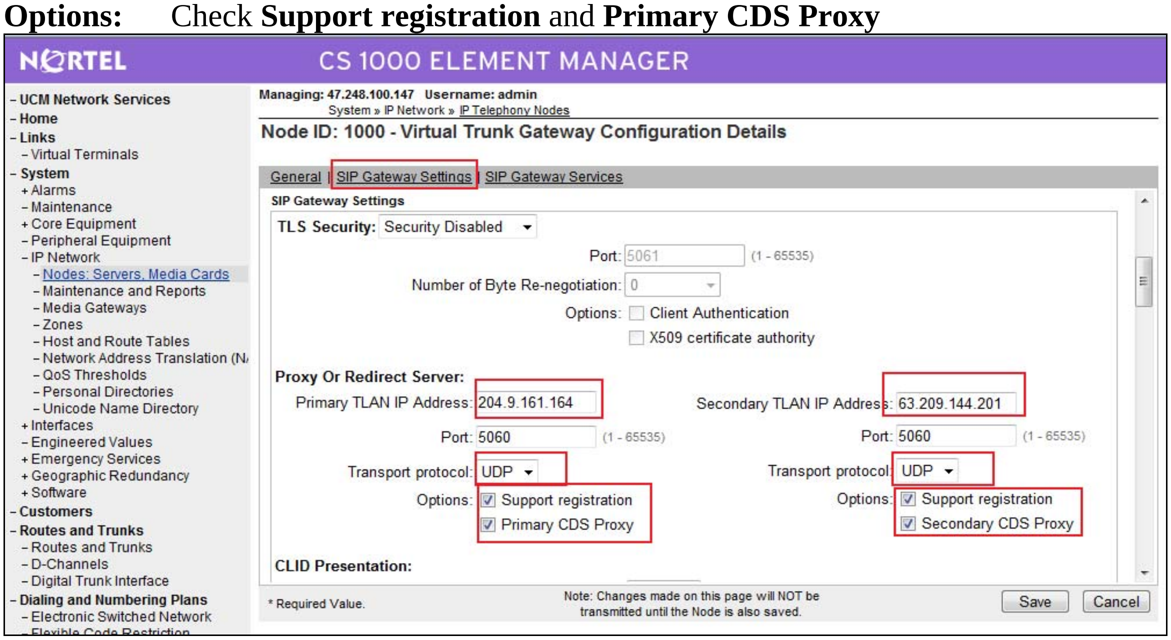

Figure 29 – Virtual Trunk Gateway Configuration Details Page.

f) Scroll down the parameters box to the SIP URI Map section.
Under 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. Click on Save.

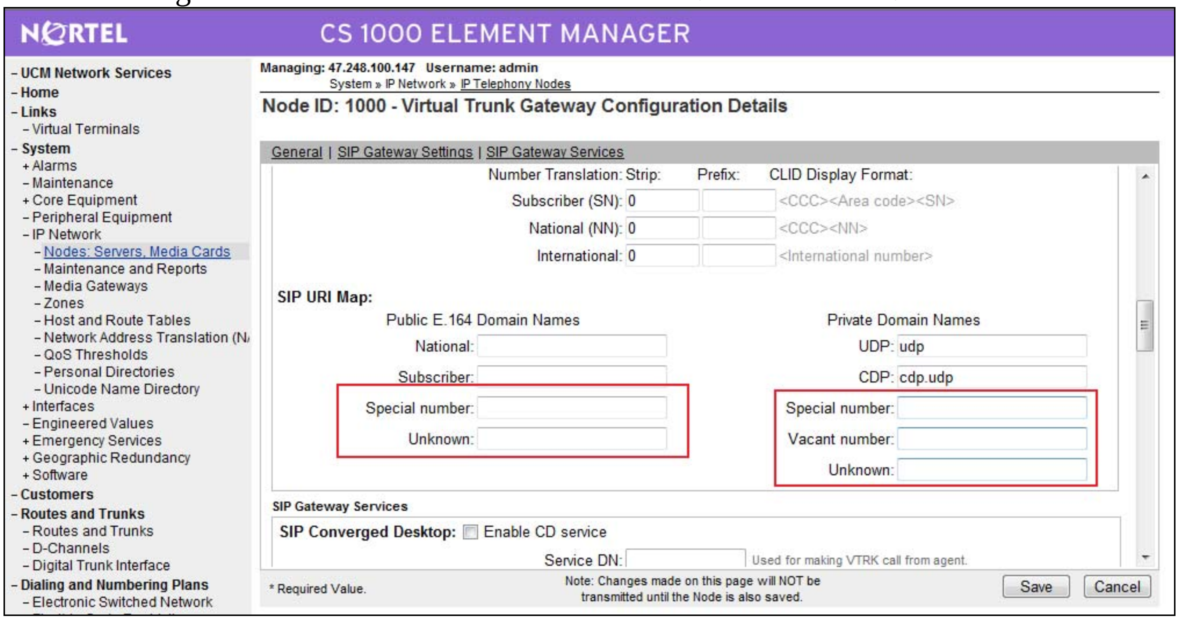

Figure 30 – Virtual Trunk Gateway Configuration Details Page.

g) **Synchronize** the new configuration (please refer to Section 4.2.4)

h) After configuration completes, on Skype Manager, the message "SIP user successfully registered at sip.skype.com" will be displayed. Please refer to Section 5 for more detail.

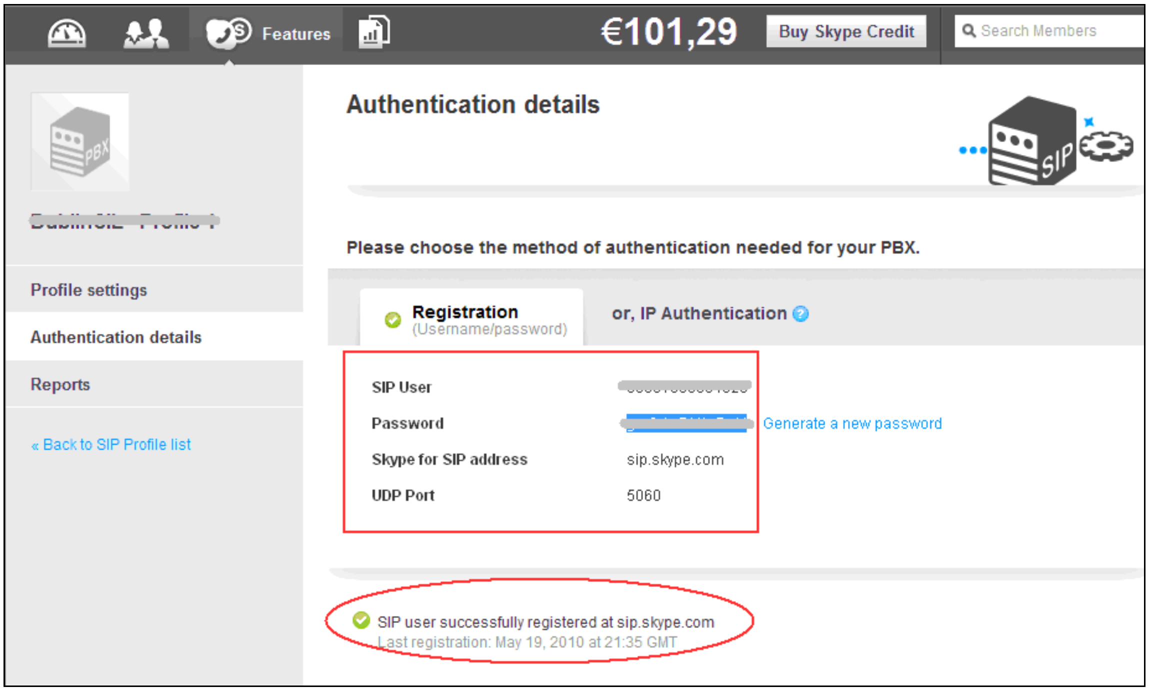

Figure 31 - CS1000 registered successfully to Skype.

4.5.3. Administer Virtual D-Channel

a) Log in UCM and EM (please refer to Section 4.1.1 for more detail)
b) Select **Routes and Trunks** -> **D-Channels** from the left pane to display the **D-Channels** screen. In the **Choose a D-Channel Number** field, select an available D-channel from the drop-down list. Click to **Add.**

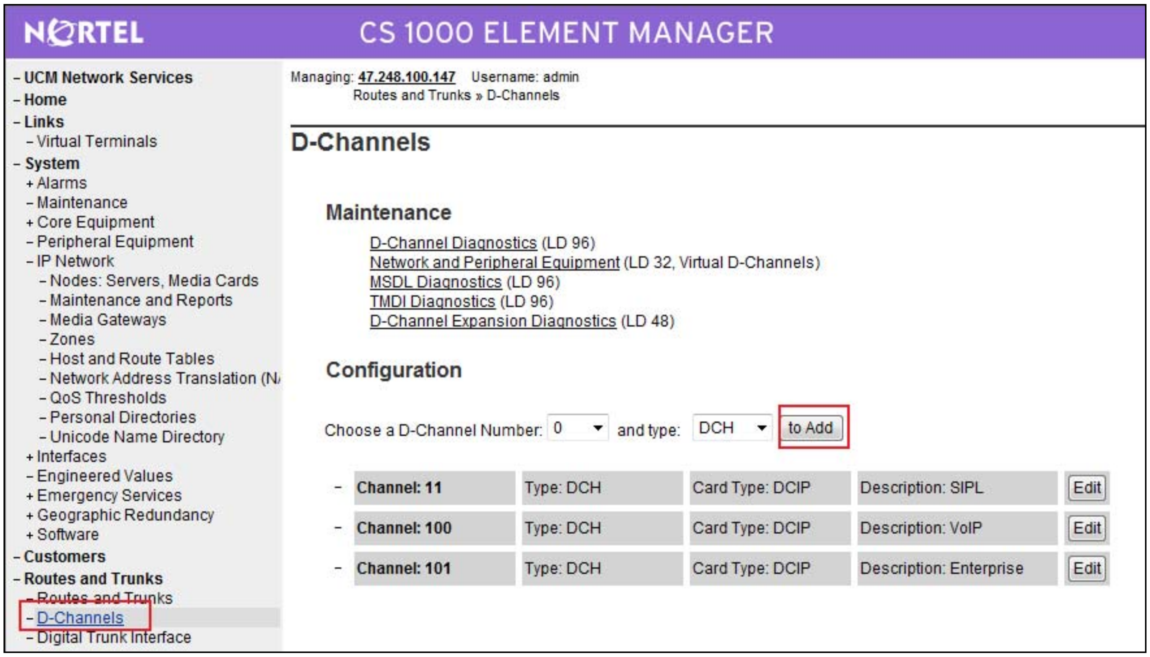

Figure 32 – D-Channels Page.

c) The D-Channels 100 Property Configuration screen is displayed next. Enter the following values for the specified fields, and retain the default values for the remaining fields.

D channel Card Type (CTYP): D-Channel is over IP (DCIP)
Designator (DES): A descriptive name
Interface type for D-channel (IFC): Meridian Meridian1 (SL1)
Release ID of the switch at the far end (RLS): 6

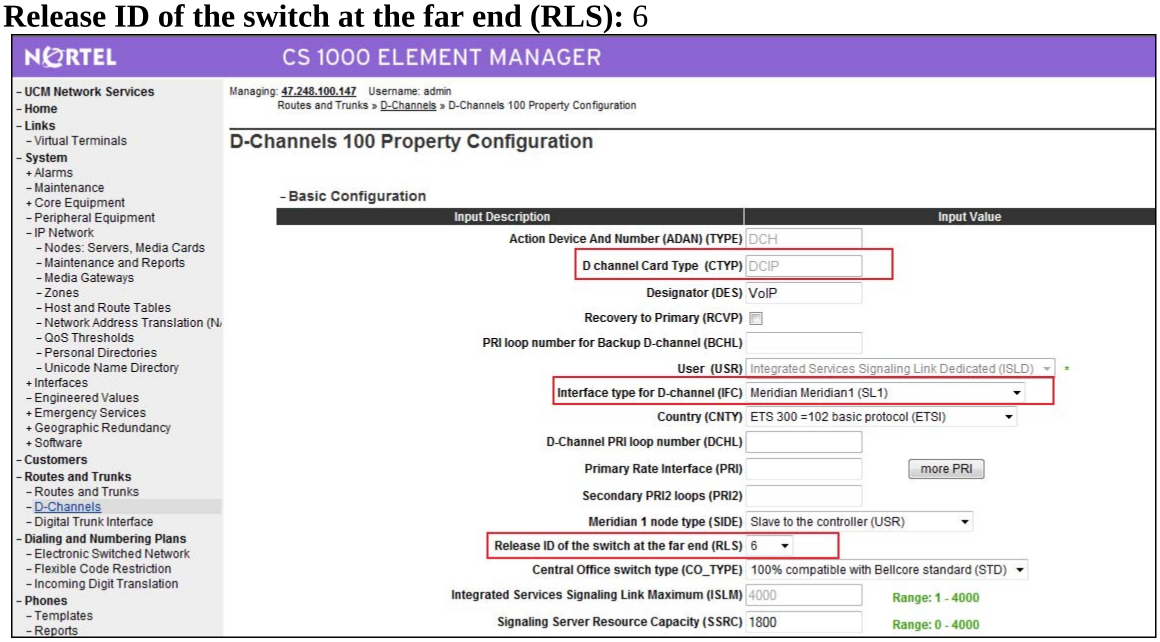

Figure 33 – D-Channels Configuration Details Page.

d) click on **Basic Options** and select the **Remote Capabilities** (**RCAP**). then enable **ND2, MWI** if CS1000 hosted voice mail will be used.

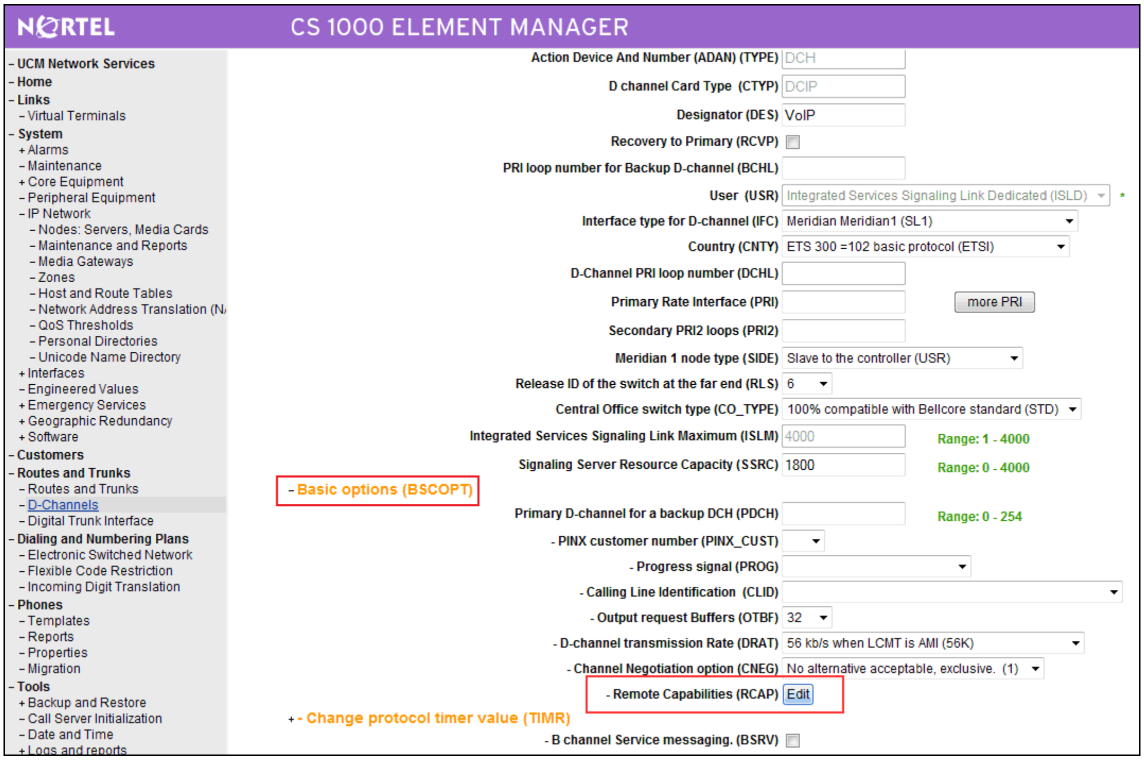

Figure 34 – D-Channels Configuration Details Page.

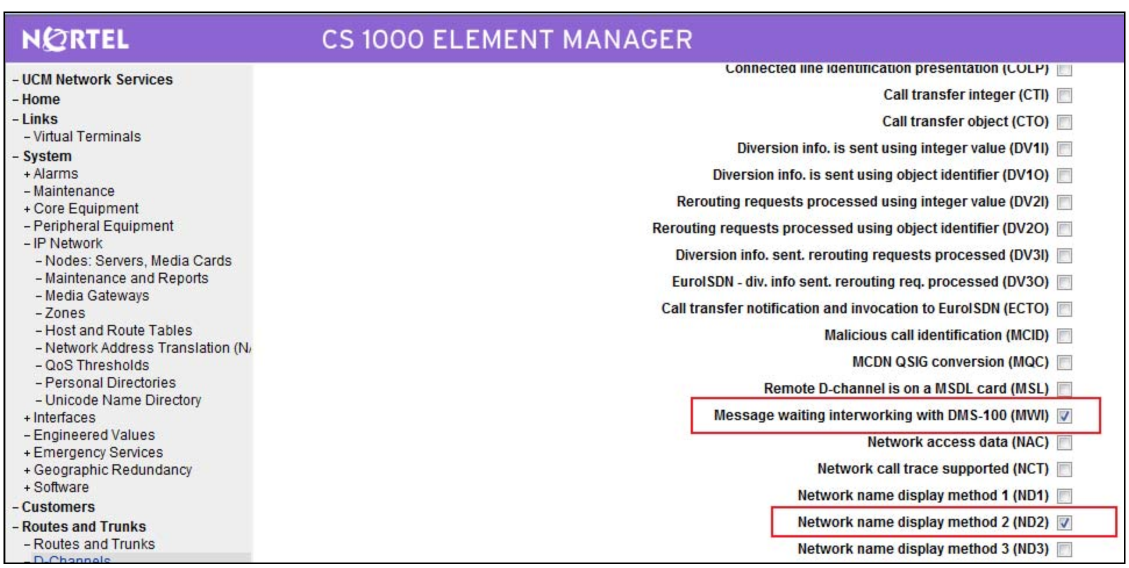

Figure 35 – D-Channels Configuration Details Page.

e) Click **Return – Remote Capabilities** (not shown).
f) Click **Submit** (not shown).

4.5.4. Administer Virtual Super-Loop

a) Log in UCM and EM (please refer to Section 4.1.1 for more detail)
b) Select **System** -> **Core Equipments** -> **Superloops** from the left pane to display the **Superloops** screen. If Superloop does not exist, please click "**Add**" button to create a new one.

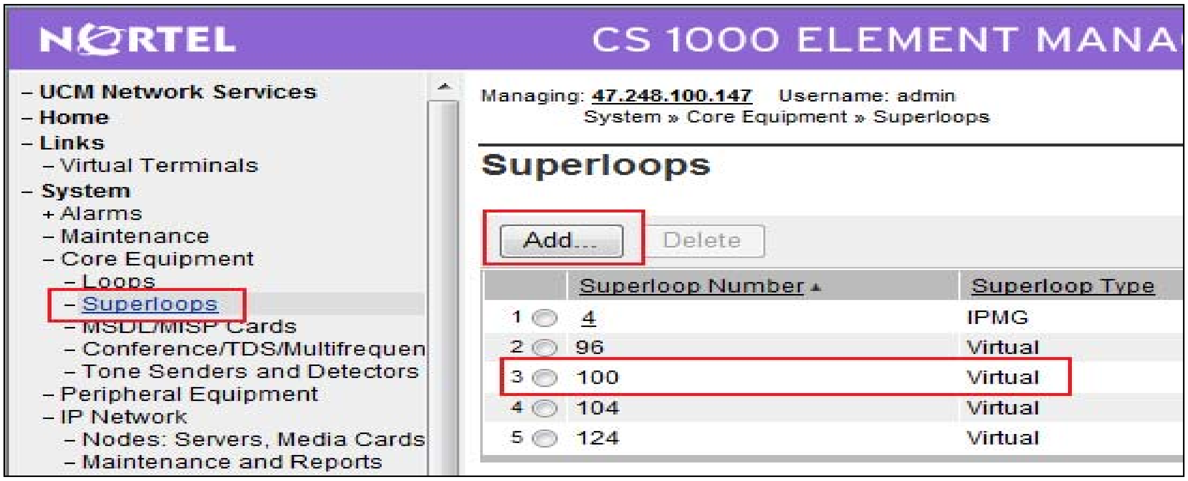

Figure 36 – Administer Virtual Super-Loop Page.

4.5.5. Administer Virtual SIP Routes

a) Log in UCM and EM (please refer to Section 4.1.1 for more detail)
b) Select **Routes and Trunks** -> **Routes and Trunks** from the left pane to display the **Routes and Trunks** screen. Next to the applicable **Customer** row, click **Add route**.

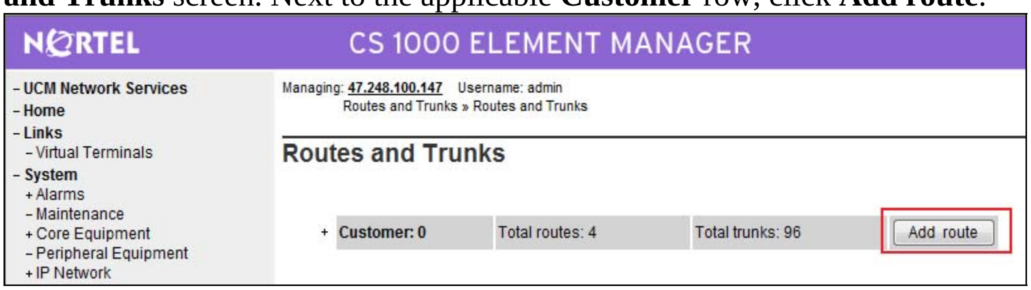

Figure 37 – Add route.

c) The **Customer 0**, New **Route Configuration** screen is displayed next. Scroll down until the **Basic Configuration** Section is displayed and enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Route Number** (ROUT): Select an available route number.
- **Designator field for trunk** (DES): A descriptive text.
- **Trunk Type** (TKTP): TIE trunk data block (TIE)
- **Incoming and Outgoing trunk** (ICOG): Incoming and Outgoing (IAO)
- **Access Code for the trunk route** (ACOD): An available access code.
- Check the field **The route is for a virtual trunk route** (VTRK), to enable four additional fields to appear.
- For the **Zone for codec selection and bandwidth management** (ZONE) field, enter 255 (created in Section 4.4.2).
- For the **Node ID of signaling server of this route** (NODE) field, enter the node number 1000 (created in Section 4.2.1).
- Select **SIP** (SIP) from the drop-down list for the **Protocol ID for the route** (PCID) field.
- Check the **Integrated Services Digital Network option** (ISDN) checkbox to enable additional fields to appear. Enter the following values for the specified fields, and retain the default values for the remaining fields. Scroll down to the bottom of the screen.
    - **Mode of operation** (MODE): Route uses ISDN Signaling Link (ISLD)
    - **D channel number** (DCH): D-Channel number 100 (created in Section 4.5.3)
    - **Network calling name allowed** (NCNA): Check the field.
    - **Network call redirection** (NCRD): Check the field.
    - **Insert ESN access code** (INAC)**:** Check the field.

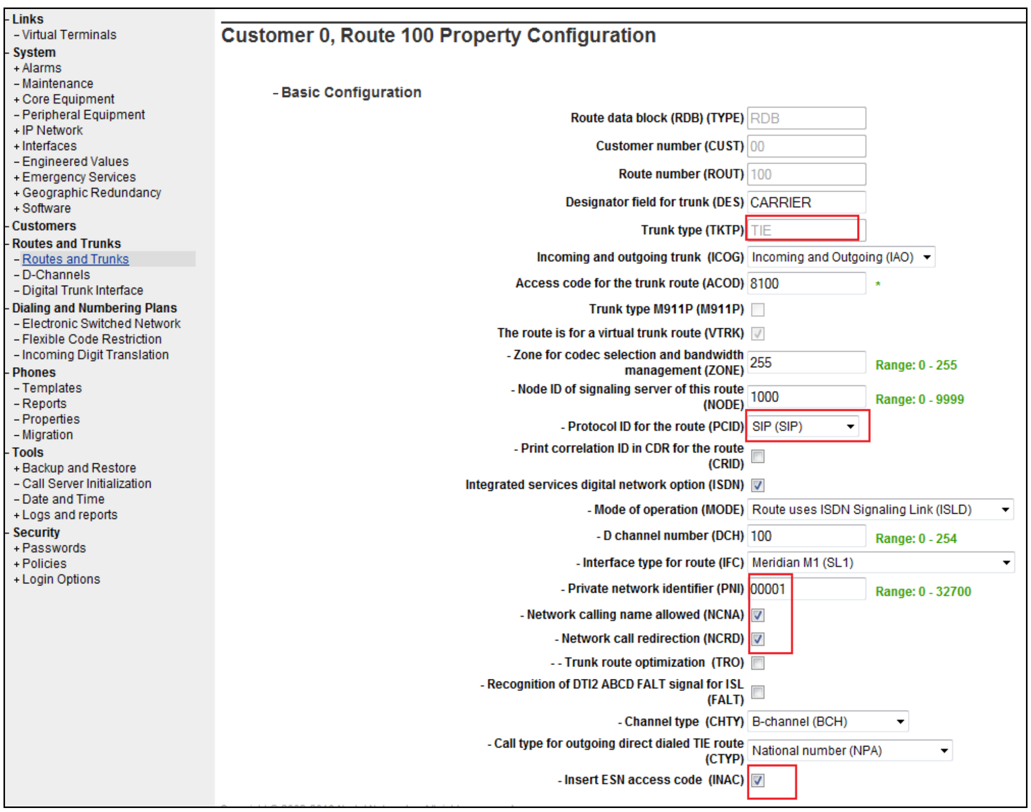

Figure 38 – Route Configuration Details Page.

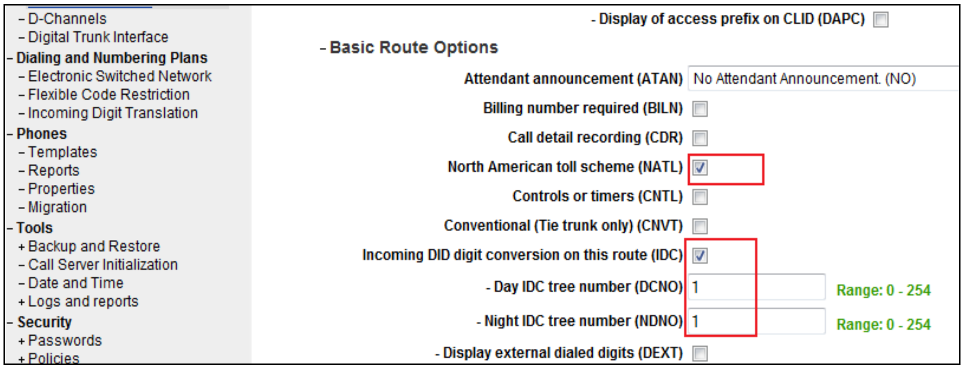

Figure 39 – Route Configuration Details Page.

d) Click **Submit** (not shown).

4.5.6. Administer Virtual Trunks

a) Log in UCM and EM (please refer to Section 4.1.1 for more detail)
b) Continue Section 4.5.5, after click **Submit**, the **Routes and Trunks** screen is displayed and updated with the newly added route. Click the **Add trunk** button next to the newly added route.

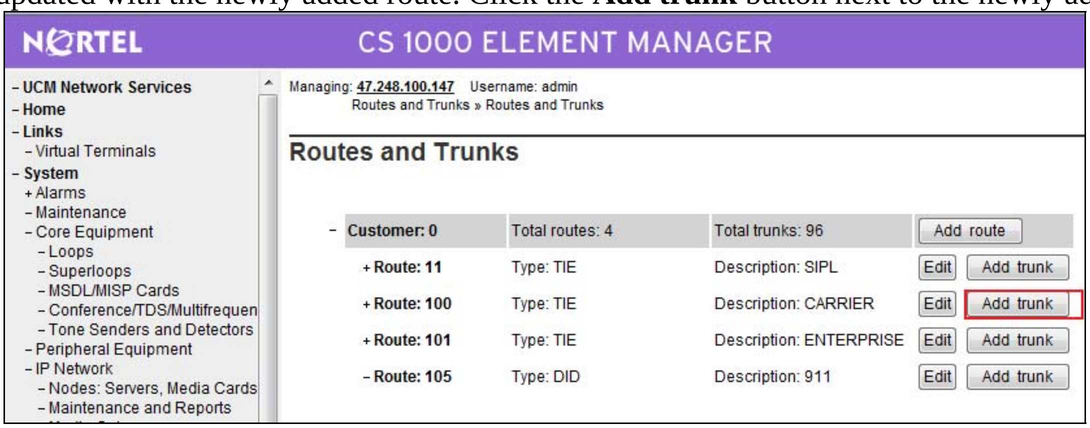

Figure 40 – Route and Trunks Page.

c) The **Customer 0, Route 15, Trunk 1 Property Configuration** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields. And then must disable Media Security (sRTP) at the trunk level by editing the **Class of Service** (CLS) at the bottom basic trunk configuration page shown in **Figure 41**.

- The Multiple trunk input number (**MTINPUT**) field may be used to add multiple trunks in a single operation, or repeat the operation for each trunk. In the sample configuration, four trunks were created.
- Trunk data block (**TYPE**): IP Trunk (IPTI)
- Terminal Number (**TN**): Available terminal number (created in Section 4.5.4)

- Designator field for trunk (**DES**): A descriptive text
- Extended Trunk (**XTRK**): Virtual trunk (VTRK)
- Route number, Member number (**RTMB**): Current route number and starting member
- Card Density: 8D
- Start arrangement Incoming (**STRI**): IMM
- Start arrangement Outgoing (**STR**O): IMM
- Trunk Group Access Restriction (**TGAR**): Desired trunk group access restriction level
- Channel ID for this trunk (**CHID**): An available starting channel ID

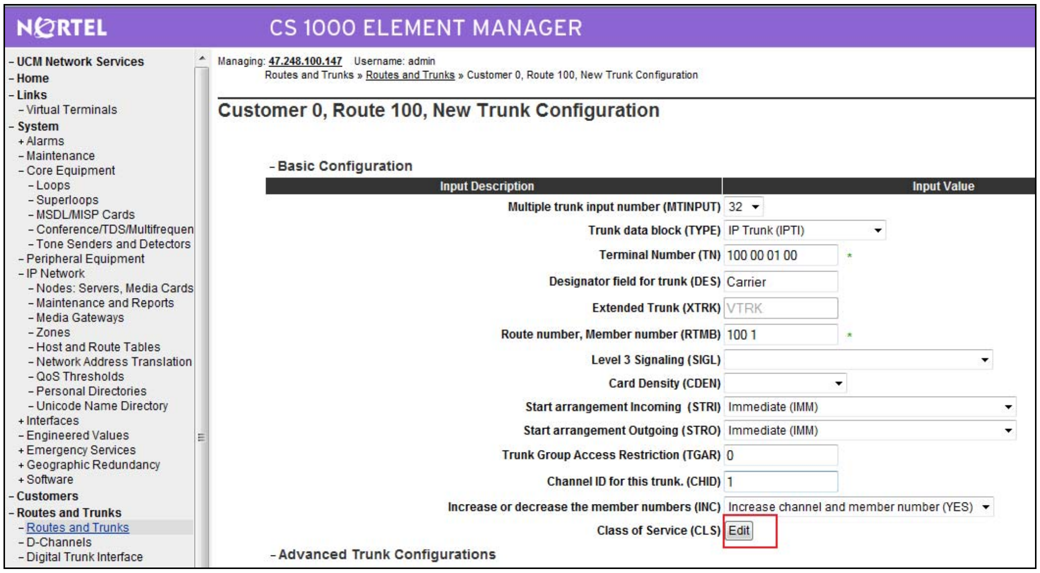

Figure 41 – new Trunk Configuration Details Page.

d) For **Media Security**, select **MSNV**. Enter the remaining values for the specified fields as shown in the following figure. Scroll down to the bottom of the screen and click **Return Class**

of Server and then click **Save** (not shown).

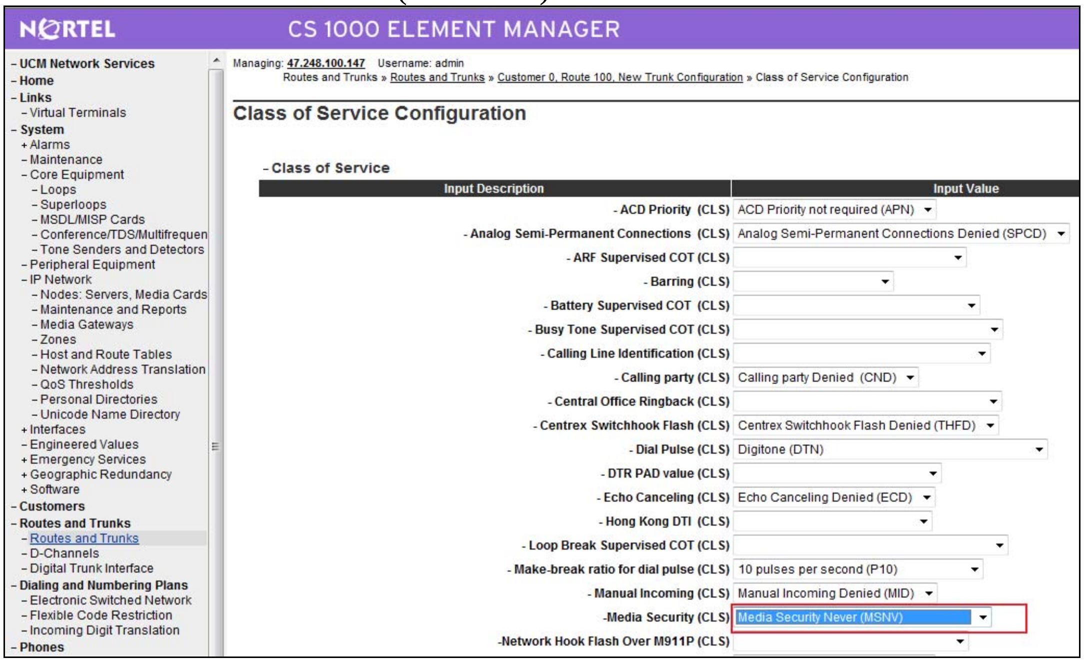

Figure 42 – Class of Service Configuration Details Page.

4.6. Administer Dialing Plans

4.6.1. Digit Manipulation Block (DMI) for Inbound Call (DMI 7)

a) Log in UCM and EM (please refer to Section 4.1.1 for more detail)
b) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen. Select **Digit Manipulation Block** (DGT).

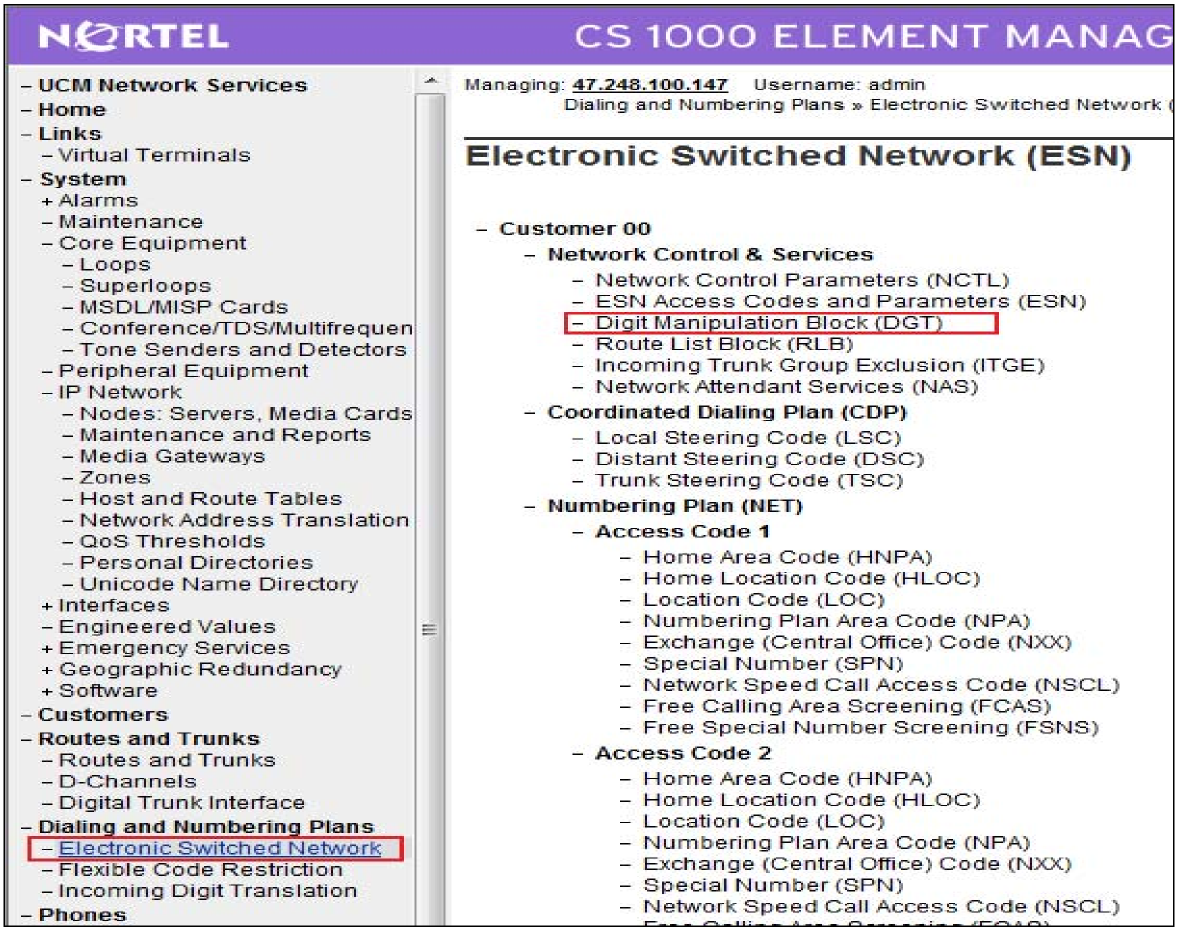

Figure 43 –ESN Configuration Details Page.

c) In the Choose a DMI Number field, select an available DMI from the drop-down list and click **to Add**

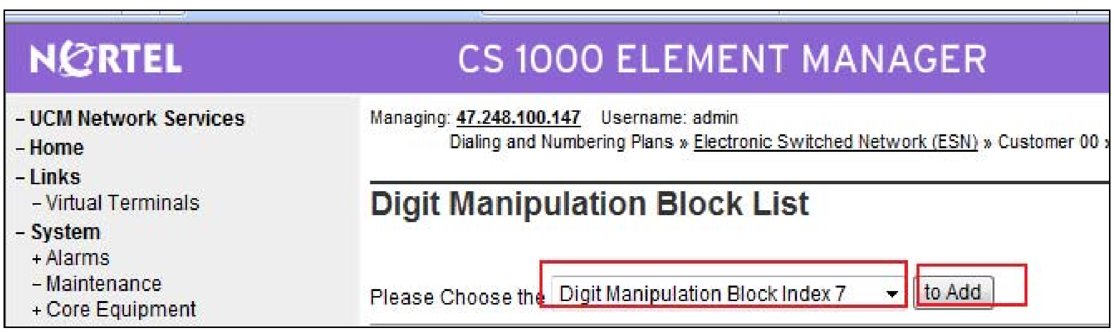

Figure 44 –Add a DMI.

d) Enter 7 for the **Number of leading digits to be Deleted** (Del) field and select **NPA** for the **Call Type to be used by the manipulated digits** (CTYP) and then click **Submit.**

Figure 45 – DMI Configuration Details Page.

4.6.2. Digit Manipulation Block (DMI) for Outbound Call (DMI 25)

a) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen. Select **Digit Manipulation Block** (DGT) as above.

b) In the Choose a DMI Number field, select an available DMI from the drop-down list and click **to Add** button.

Figure 46 – Add a DMI.

d) Enter 0 for the **Number of leading digits to be Deleted** (Del) field and select **NPA** for the **Call Type to be used by the manipulated digits** (CTYP) and then click **Submit.**

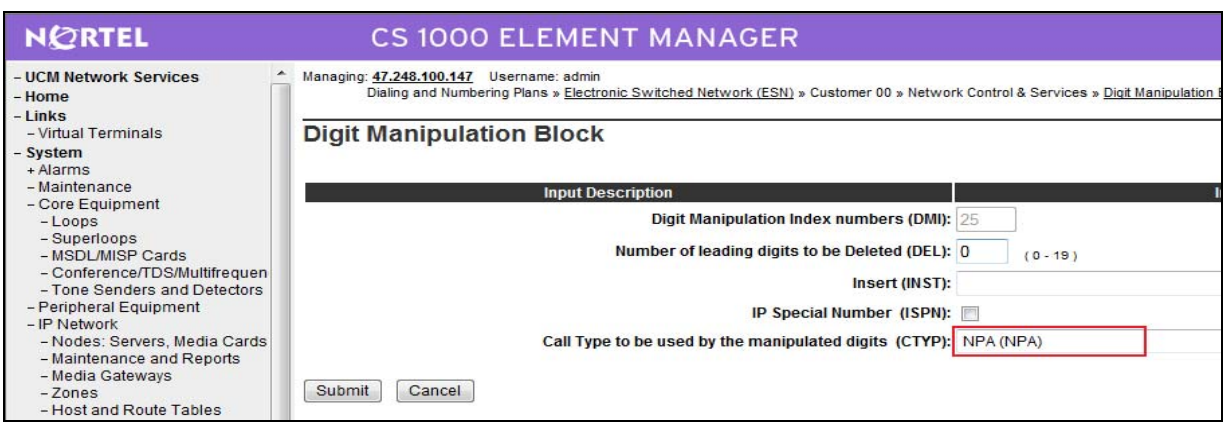

Figure 47 – DMI Configuration Details Page.

4.6.3. Route List Block (RLB) for National Call (RLB 25)

a) Log in UCM and EM (please refer to Section 4.1.1 for more detail)

b) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen. Select **Route List Block** (RLB).

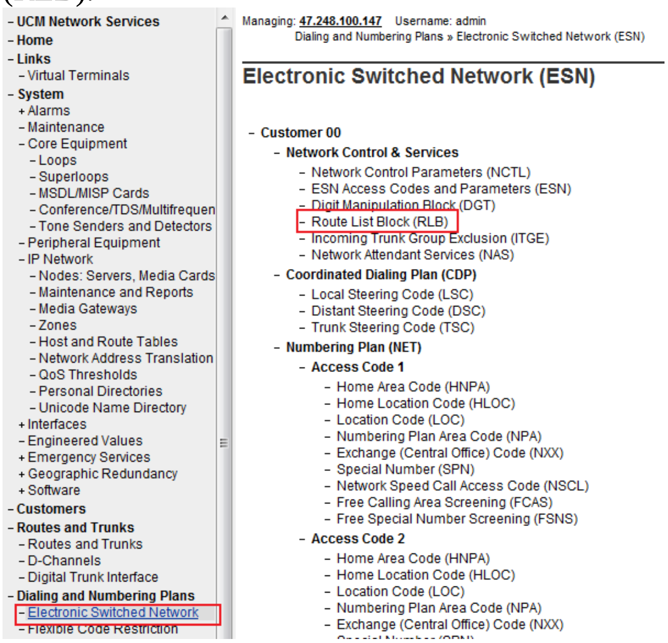

Figure 48 – ESN Configuration Details Page.

c) Select an available value to and click **to Add** (in this case is 25)

Figure 49 – Add a Route List Blocks.

d) Enter the following values for the specified fields, and retain the default values for the remaining fields. Scroll down to the bottom of the screen, and click **Submit.**

Route number (ROUT) : 100 (created in Section 4.5.5)

Digit Manipulation Index (DMI): 25 (created in Section 4.6.2)

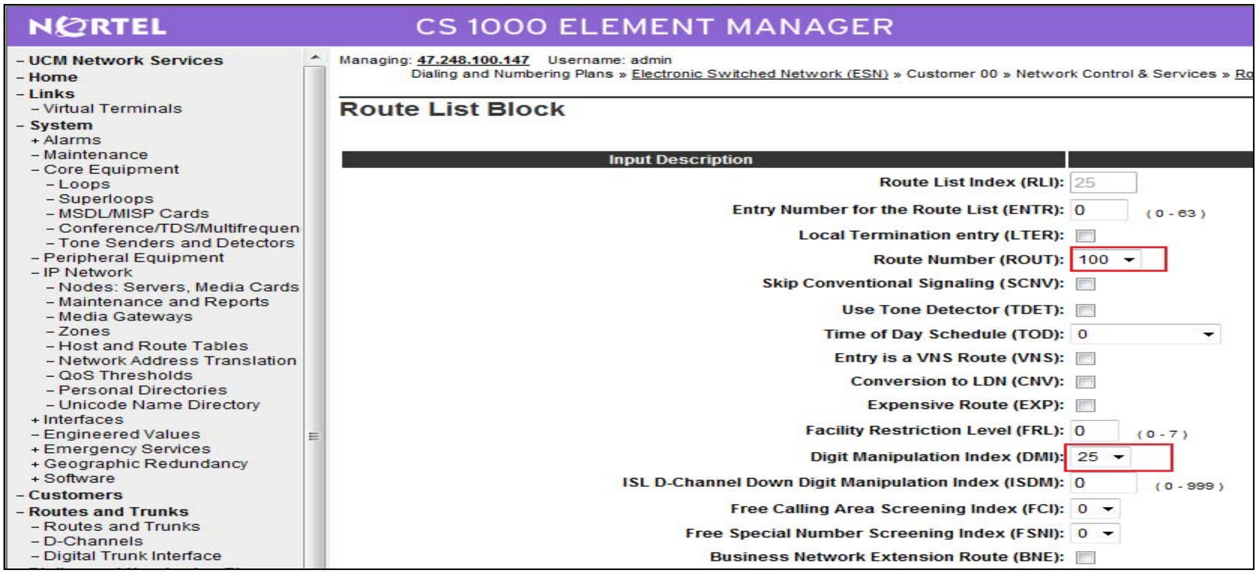

Figure 50 – Route List Blocks Configuration Details Page.

4.6.4. Route List Block (RLB) for International Call (RLB 26)

a) Log in UCM and EM (please refer to Section 4.1.1 for more detail)
b) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen. Select **Route List Block** (RLB) as **Figure 48**.
c) Select an available value to and click **to Add** (in this case is 26)

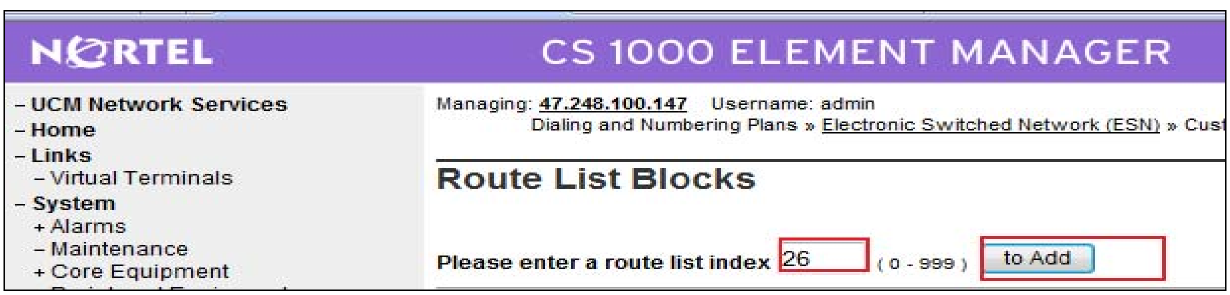

Figure 51 – Add a Route List Blocks.

d) Enter the following values for the specified fields, and retain the default values for the remaining fields. Scroll down to the bottom of the screen, and click **Submit.**

- **Route number** (ROUT) : 100 (created in Section 4.5.5)
- **Digit Manipulation Index** (DMI): 25 (created in Section 4.6.2)
- **Facility Restriction Level** (FRL): 7

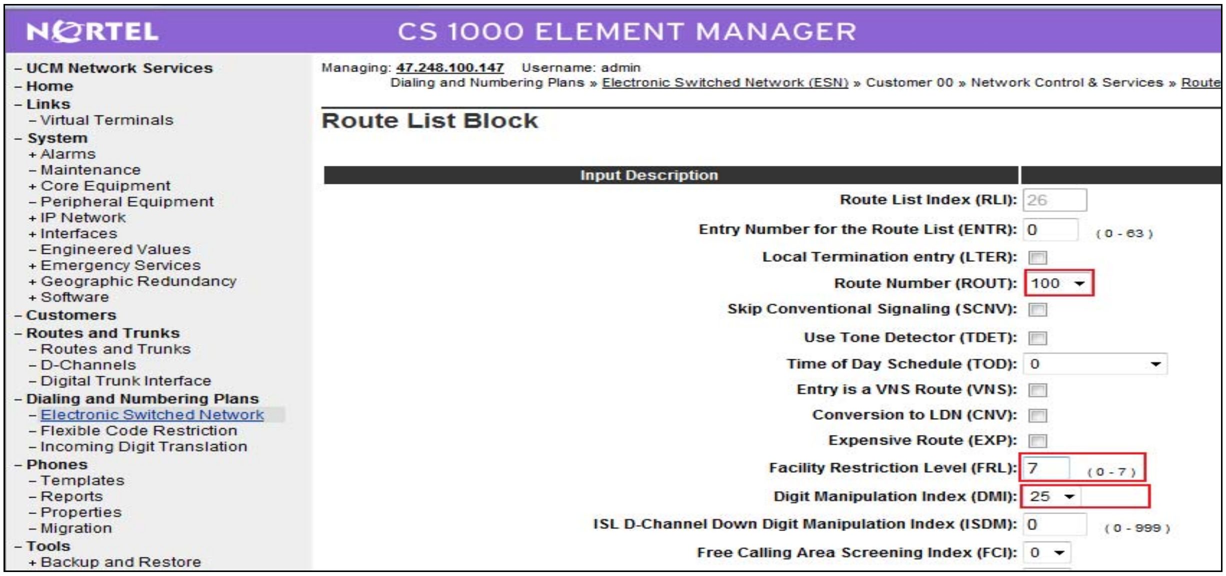

Figure 52 – Route List Blocks Configuration Details Page.

4.6.5. Inbound Call

This section describes the steps for receiving the calls from the online numbers 1315 xxx xxxx

a) Log in UCM and EM (please refer to Section 4.1.1 for more detail)
b) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen. Select **Local Steering Code** (LSC).

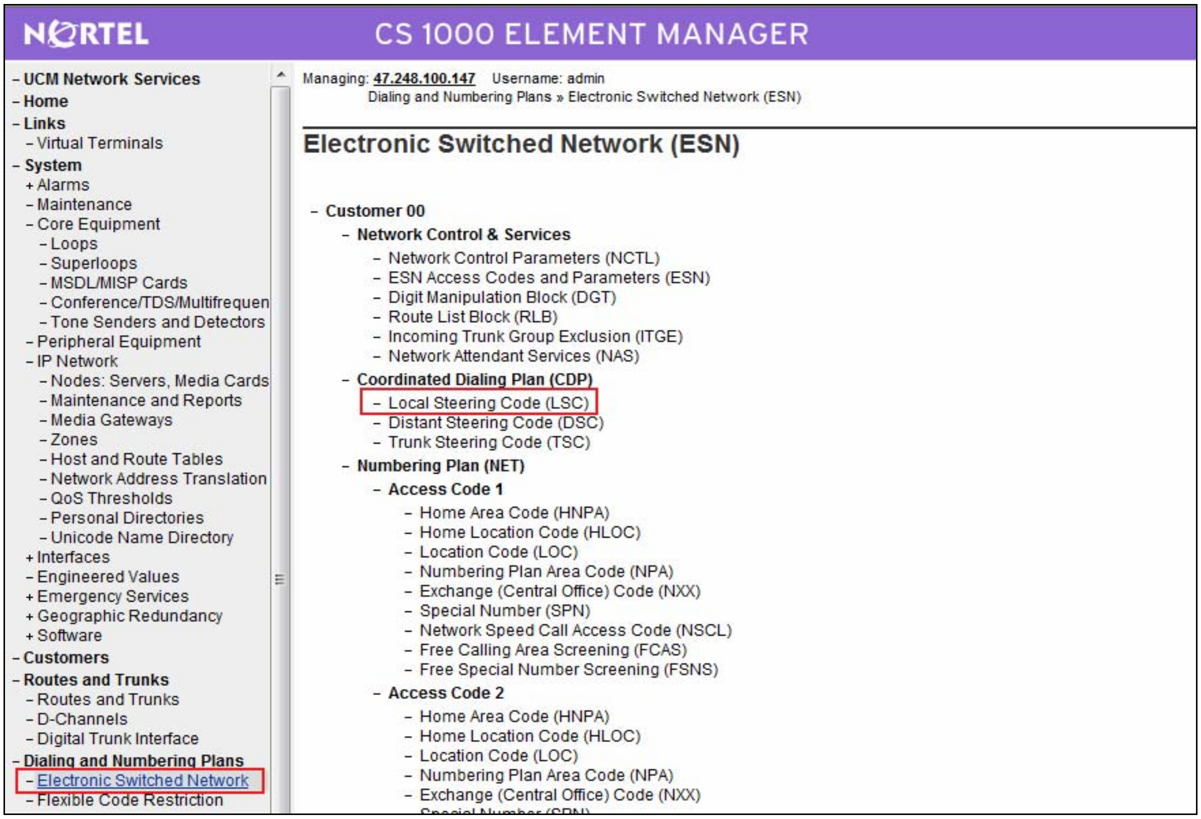

Figure 53 – ESN Configuration Page.

c) Enter 131 for the LSC field and click **to Add**

Figure 54 – Add a LSC.

d) In the Choose a DMI field, select an available DMI from the drop-down list. In this case, it's DMI 7. (created in Section 4.6.1) and then click **Submit**

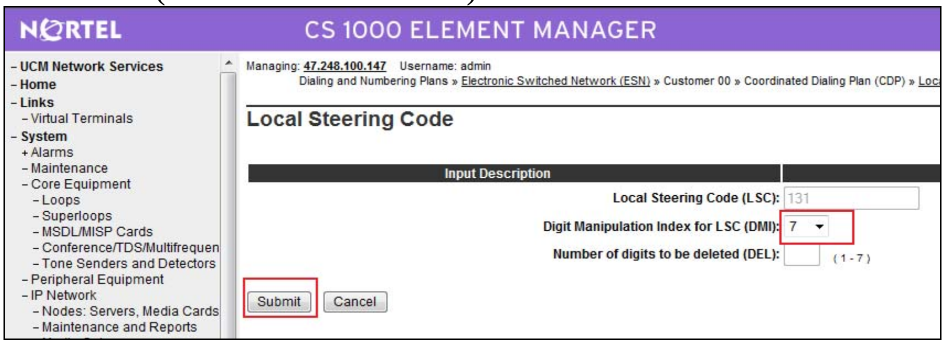

Figure 55 – LSC Configuration Details Page.

4.6.6. Inbound Call – Configure IDC to receive the call on an existing phone.

To receive the call to the online number 13157914457 on a phone having DN 3111, we can configure IDC as follows.

a) Configure FCR in Customer by ld 15. This section prints FCR configuration details.

>ld 21
PT1000

REQ: prt
TYPE: fcr
TYPE FCR_DATA
CUST 0

TYPE FCR_DATA
CUST 00
NFCR **YES**
  MAXT **100**
  OCB1 255
  OCB2 255
  OCB3 255
IDCA **YES**
  DCMX **100**

b) Configure IDC by ld 49. This section prints IDC configuration details.

>ld 49
DGT000
MEM AVAIL: (U/P): 103093188   USED U P: 481334 77827   TOT: 103652349
DISK SPACE NEEDED: 60 KBYTES
REQ prt
TYPE idc
CUST 0
DCNO

DCNO   **1** <----- this number is configured in Rout 100,

******** Note *********
LD 16
ROUT 100
....
IDC YES
DCNO 1
NDNO 1 *
*********End Note *********
SDID NO
IDGT CDGT
13157914457 3111

MEM AVAIL: (U/P): 103093188 USED U P: 481334 77827 TOT: 103652349
DISK SPACE NEEDED: 60 KBYTES
REQ

4.6.7. Outbound National Call

This section describes the steps for the outbound calls to US and CA.

a) Log in UCM and EM (please refer to Section 4.1.1 for more detail)
b) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen. Select **Special Number** (SPN).

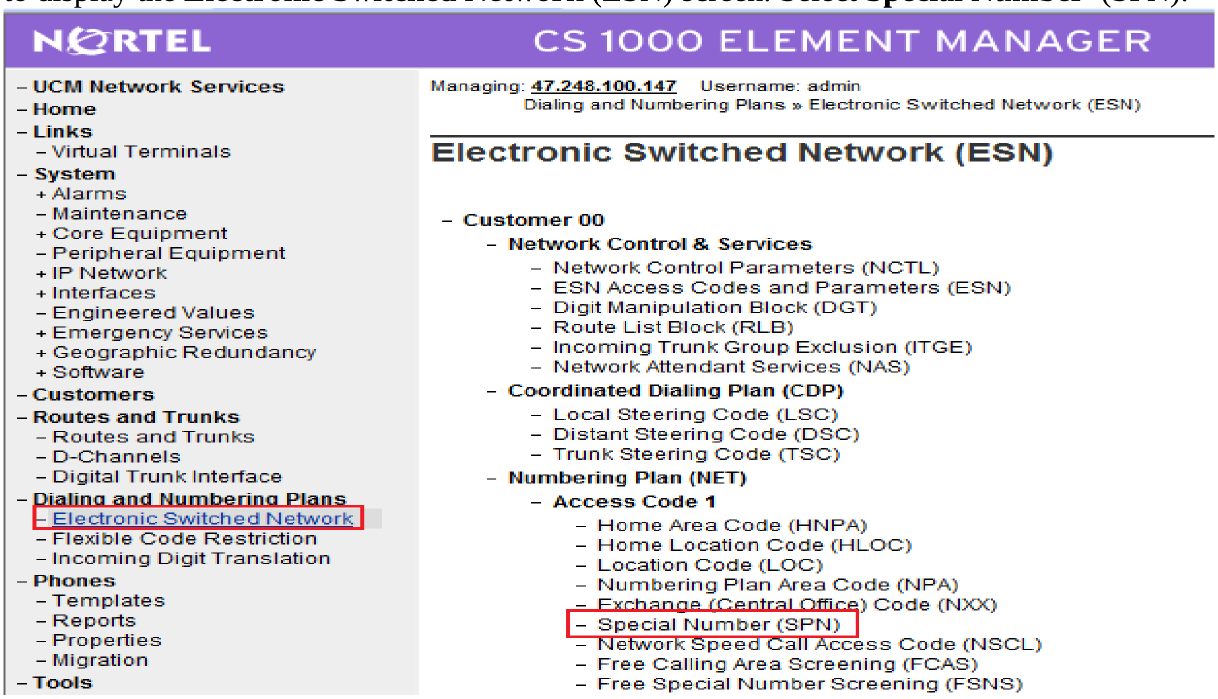

Figure 56 – ESN Configuration Page.

c) Enter a country code (1 for US, CA) for SPN and click “**to Add**”

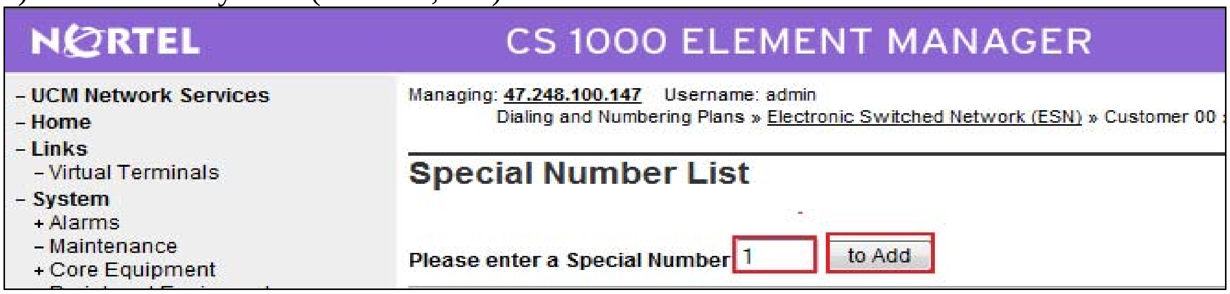

Figure 57 – Add a SPN.

d) Enter the following values for the specified fields, and retain the default values for the remaining fields. Scroll down to the bottom of the screen, and click **Submit.**

- **Flexible Length** (FLEN) : 11
- **Route List Index** (RLI) : 25 (created in Section 4.6.3)

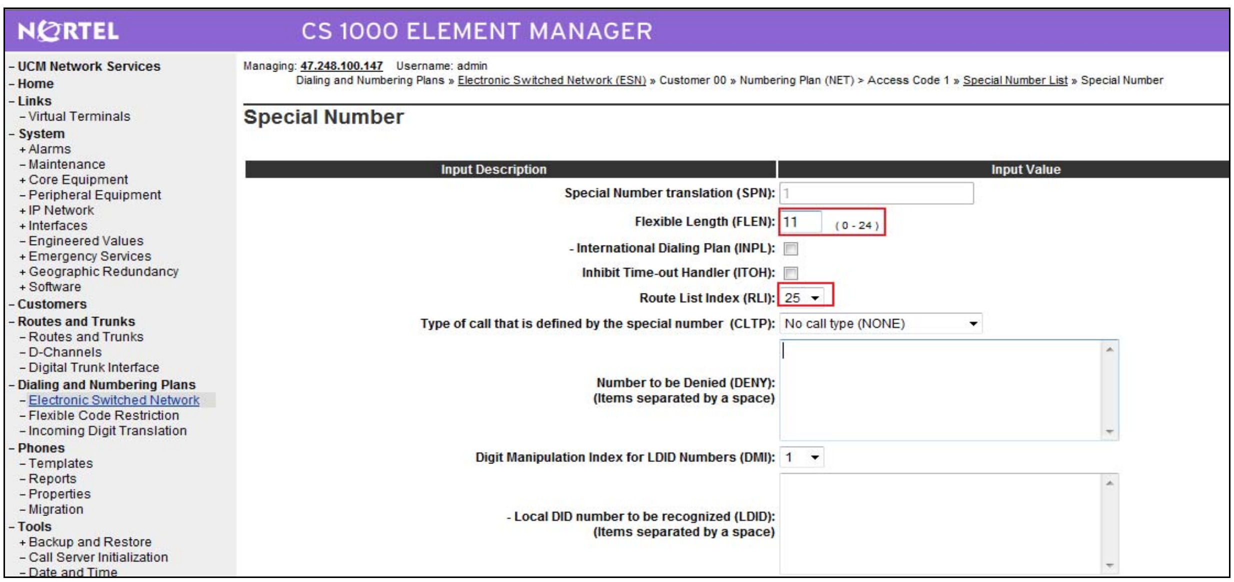

Figure 58 – SPN Configuration Details Page.

4.6.8. Outbound International Call

This section describes the steps for the international calls.

a) Log in UCM and EM (please refer to Section 4.1.1 for more detail)
b) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen. Select **Special Number** (SPN) as **Figure 56**.

c) Enter 0 for SPN and click “**to Add**”

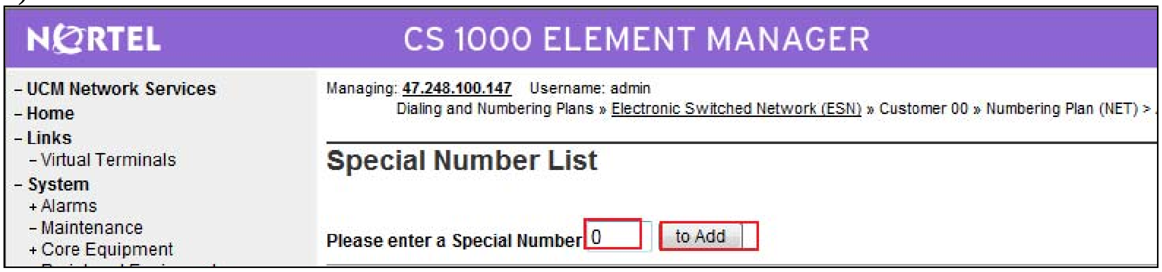

Figure 59 – Add a SPN.

d) Enter the following values for the specified fields, and retain the default values for the remaining fields. Scroll down to the bottom of the screen, and click **Submit.**

- **Flexible Length** (FLEN) : 13
- **Route List Index** (RLI) :
    - o Select 26 (created in Section 4.6.4) if user wants to enable International call restriction feature (need to continue the steps in Section 4.6.8).
    - o Select 25 (created in Section 4.6.3) to allow all phones able to make international call.

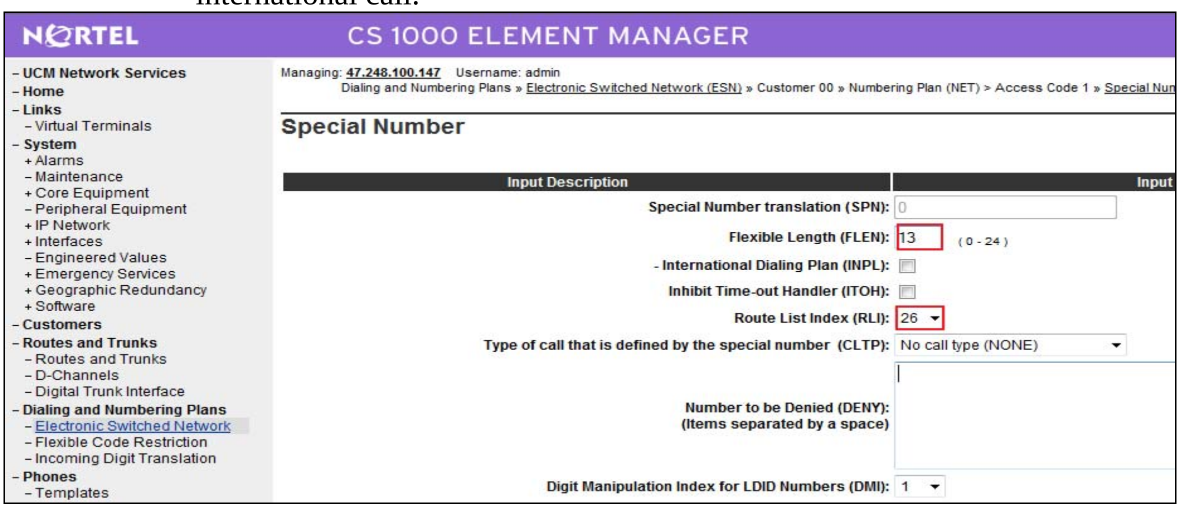

Figure 60 –SPN Configuration Details Page.

4.6.9. Outbound International Call Restriction

Before starting this section, please log in to CS overlay. Please refer to Section 4.1.2 for more detail.

a) FCR configuration:

Create a FCR by using overlay 49 and then apply to NCOS of the set

>ld 49

REQ **new**
TYPE **fcr**
CUST **0**
CRNO **7**
INIT **Alow**
ALOW
DENY
BYPS

b) NCOS configuration: change FRL in NCTL to 7.

>ld 87
REQ chg
CUST 0
FEAT **nctl**
SOHQ
OHTL
SCBQ
CBTL
RANE
RANC
NCOS 7
EQA
FRL 7
RWTA
NSC
OHQ
CBQ
SPRI
MPRI
PROM
NCOS
TOHQ

c) Change NCOS value of the phone to 7 to allow this phone to be able to make International calls, any phone has NCOS value lower than 7 will be blocked.

>ld 11
SL1000
MEM AVAIL: (U/P): 103093759 USED U P: 463488 77694 TOT: 103634941
DISK SPACE NEEDED: 60 KBYTES
TNS AVAIL: 32596 USED: 171 TOT: 32767

REQ: chg
TYPE: 2050pc
TN 96 0 10 0
ECHG yes
ITEM **ncos 7**
ITEM

4.7. Phone Configuration

Before starting this section, please log in to CS overlay. Please refer to Section 4.1.2 for more detail.

4.7.1. Calling Line Identification Entries (CLID)

Create a CLID to associate with the online numbers 1315791xxxx by using overlay 15

>ld 15
MEM AVAIL: (U/P): 103094302 USED U P: 463288 77351 TOT: 103634941
DISK SPACE NEEDED: 59 KBYTES
REQ: chg
TYPE: net

TYPE NET_DATA
CUST 0
OPT
AC2
FNP
CLID **yes**
SIZE
INTL
ENTRY **25**
HNTN **131**
ESA_HLCL
ESA_INHN
ESA_APDN
HLCL **5791**
DIDN **Yes**

4.7.2. IP Phone creation

Create a phone associated with the above CLID by using overlay 11

>ld 11
SL1000
MEM AVAIL: (U/P): 103094287 USED U P: 463288 77366 TOT: 103634941
DISK SPACE NEEDED: 59 KBYTES
TNS AVAIL: 32597 USED: 170 TOT: 32767

REQ: new
TYPE: 2050pc
TN 96 0 10 0
DES Test
CUST 0
NUID
NHTN
KEM
ZONE **10**

ERL
ECL
FDN
TGAR
LDN
NCOS **7 <---** this phone can make outbound international call.
RNPG
SSU
SCPW
SGRP
SFLT
CAC_MFC
CLS unr
HUNT
SCI
PLEV
DANI
AST
IAPG
MLWU_LANG
MLNG
DNDR
KEY 00 SCR 4460 **25**

4.7.3. Outbound Caller ID Restriction to PSTN

```
>ld 11
SL1000
MEM AVAIL: (U/P): 103093759    USED U P: 463488 77694    TOT: 103634941
DISK SPACE NEEDED: 60 KBYTES
TNS                    AVAIL: 32596    USED:   171    TOT: 32767

REQ: chg
TYPE: 2050pc
TN   96 0 10 0
ECHG yes
ITEM cls CLBA
ITEM
```

- (CLBD) Deactivate Calling Party Number and Name per-line blocking
- CLBA Activate Calling Party Number and Name per-line blocking.
- (DDGA) DDGD   (Allow) deny DN to be displayed on other set.
- (NAMA) NAMD   (Allow) deny name to be displayed on other set.

4.8. Configure Voicemail System (Call Pilot) on CS1000

In this section, this application note assumes that the basic configuration has already been administered. The below procedures describe the configuration details of Avaya Communication Server 1000 and Callpilot. For further information on Avaya Communications Server 1000, please consult reference in Section 9.

4.8.1. Configuration Details on CallPilot Manager

a. Log in to Callpilot: http://IP of callpilot/cpmgr

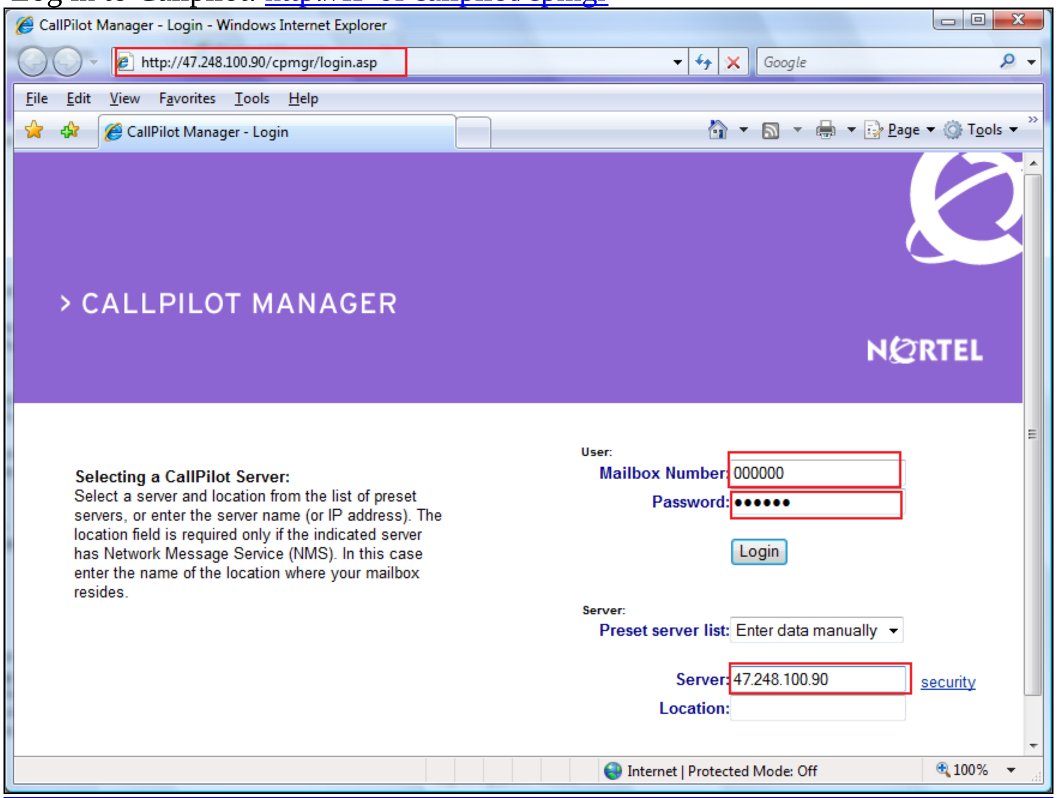

Figure 61 –Log in Callpilot.

b. Select **Configuration Wizard**.

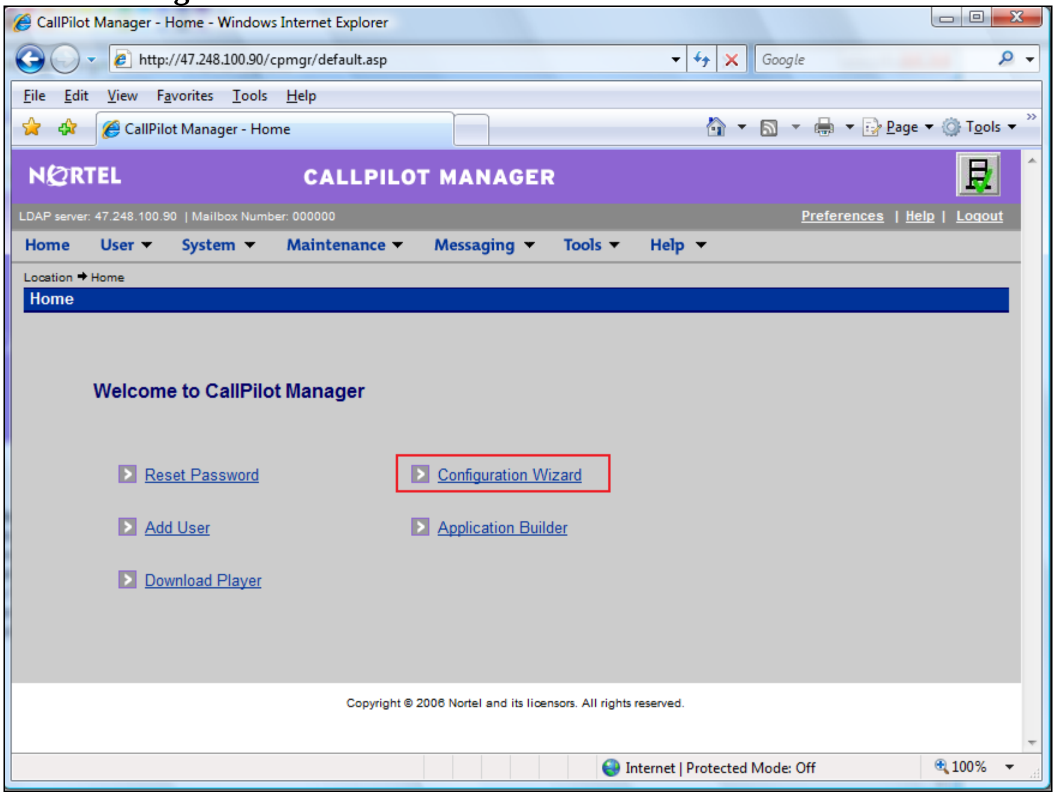

Figure 62 –Select Configuration Wizard.

c. Click **NEXT** and then select "**CallPilot Individual Feature Configuration (Express Mode**)" and click **NEXT**.

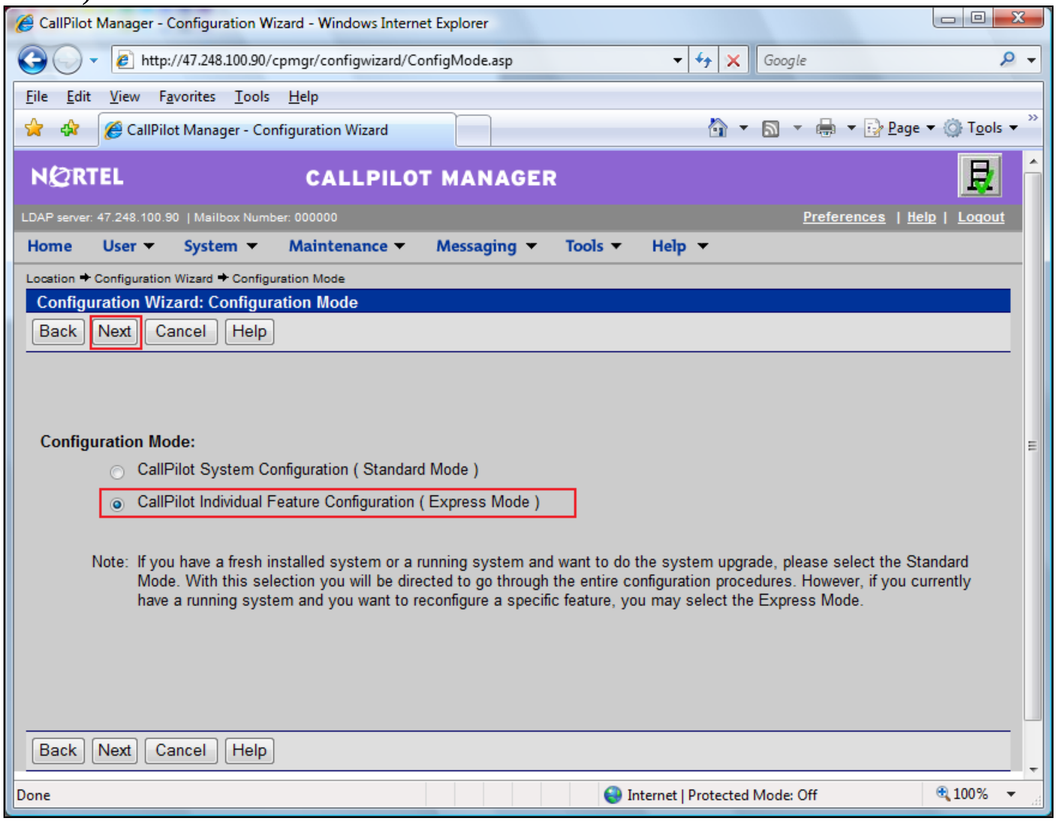

Figure 63 – Callpilot Configuration Details Page.

d. Select **Switch Configuration** and click **Next**.

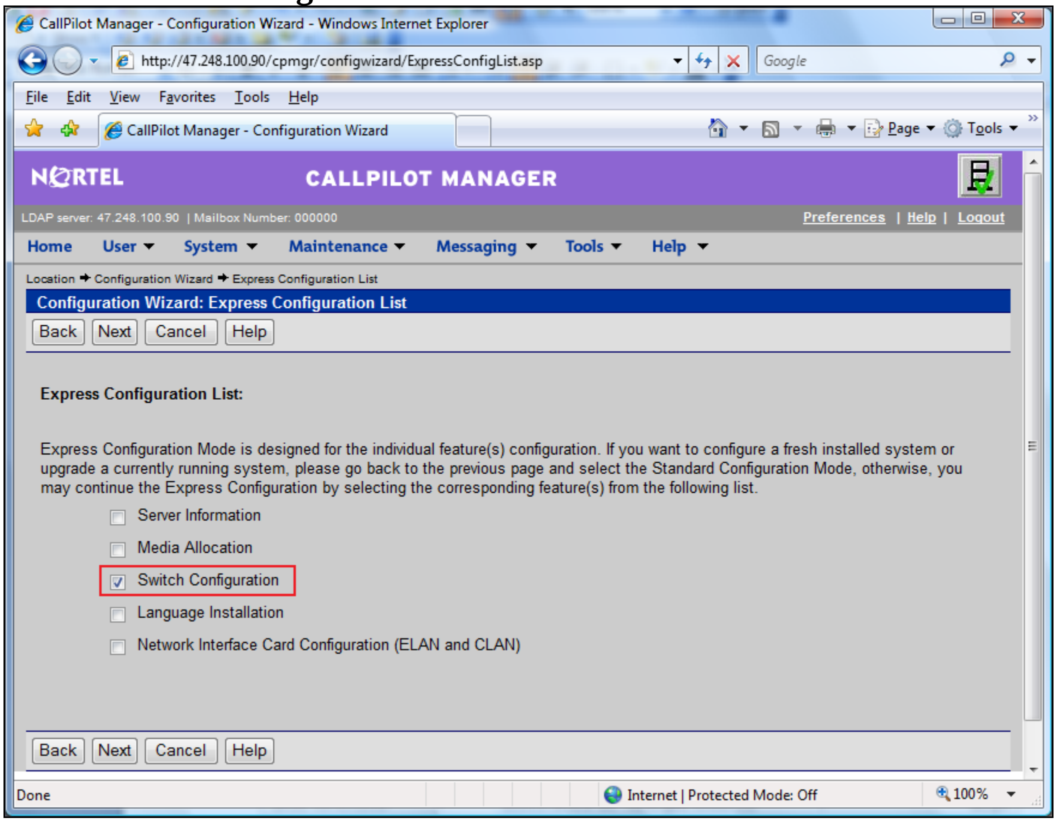

Figure 64 – Callpilot Configuration Details Page.

e. Enter CS1000 Call Server IP address and create Multimedia channels for communication between CS1000 and Callpilot system and click **Next.**

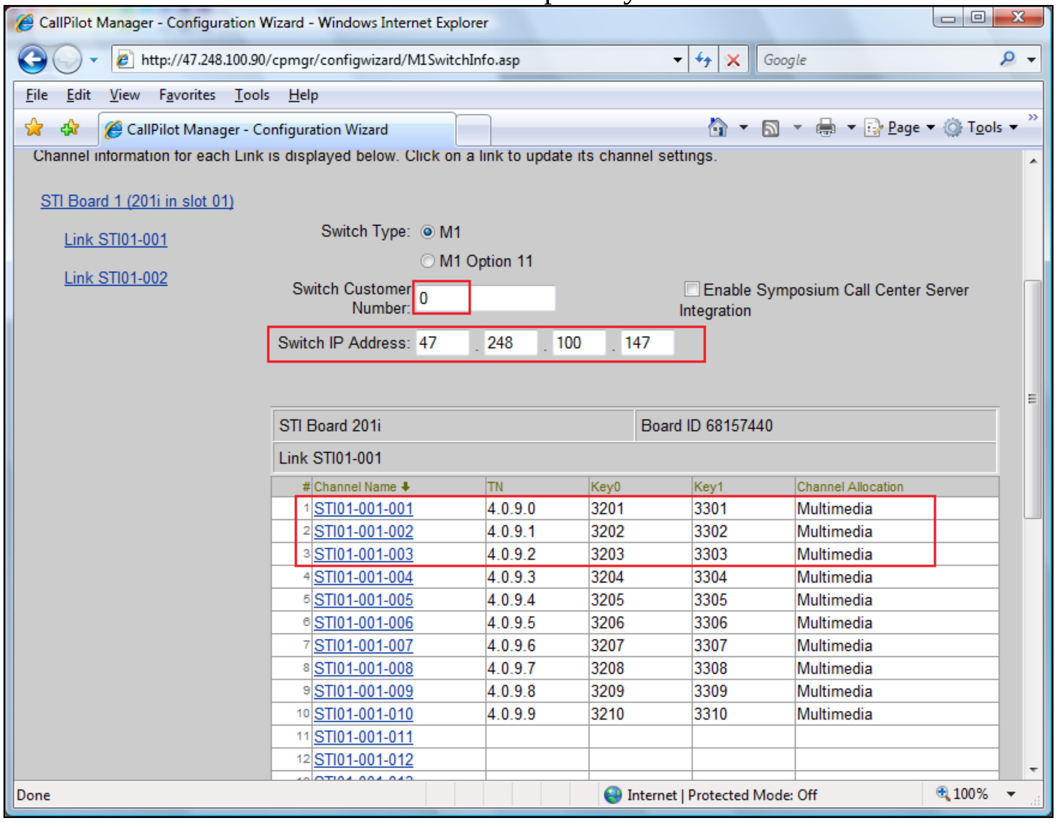

Figure 65 – Callpilot Configuration Details Page.

To get information about TNs, Key0 and Key1, please refer to Section 4.8.2. Type of Channel would be Multimedia.

f. To have a voice mailbox number, please click **New** and enter an SDN. This SDN number would be CDN configured on CS (refer to Section 4.8.2) and click **Next**

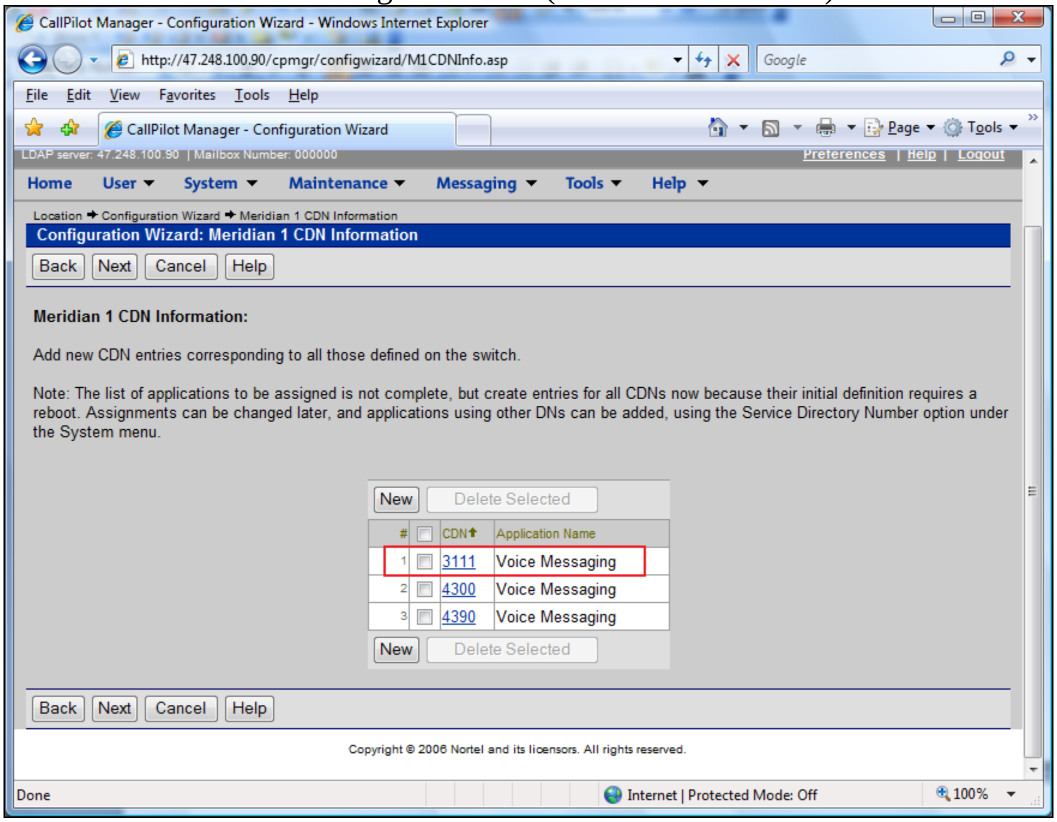

Figure 66 – Callpilot Configuration Details Page.

g. Click **Finish.**

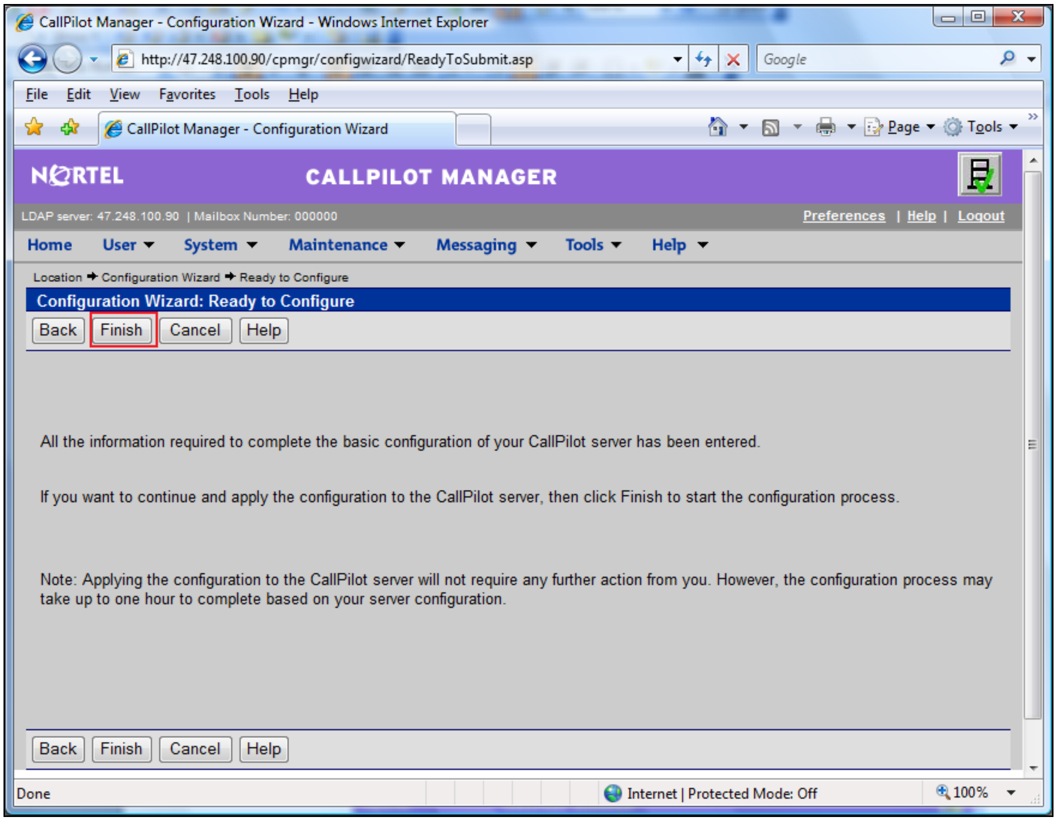

Figure 67 – Callpilot Configuration Details Page.

h. After finished, Callpilot will be out of service. Please reboot it.

i. Go to **Maintenance** pull down menu, select **Channel Monitor** to check status of the newly created multimedia channels on Call Pilot to see if the communication between

Callpilot and CS1000 has been established.

Figure 68 – Callpilot Maintenance Page.

4.8.2. Configuration Details on CS1000 Call Server

Before starting this section, please log in to CS overlay. Please refer to Section 4.1.2 for more detail.

a. Create ACD by overlay 23. Below is output of ACD 3109 after created.

>ld 23
ACD000
MEM AVAIL: (U/P): 103093188   USED U P: 481334 77827   TOT: 103652349
DISK SPACE NEEDED: 60 KBYTES
ACD DNS   AVAIL: 23997   USED: 3   TOT: 24000
REQ prt
TYPE acd
CUST 0
ACDN 3109

TYPE ACD
CUST 0
ACDN **3109**
MWC NO
DSAC NO
MAXP **10**
SDNB NO
BSCW NO
ISAP NO
AACQ NO
RGAI NO
ACAA NO

FRRT
SRRT
NRRT
FROA NO
CALP POS
ICDD NO
NCFW
FNCF NO
CWTT NONE
HMSB YES
ACPQ NO
FORC NO
RTQT 0
SPCP NO
OBTN NO
RAO NO
CWTH 1
NCWL NO
BYTH 0
OVTH 2047
TOFT NONE
HPQ NO
OCN NO
OVDN
IFDN
OVBU LNK LNK LNK LNK
EMRT
MURT
RTPC NO
NRAC NO
RAGT 4
DURT 30
RSND 4
FCTH 20
CRQS 100
CCBA NO
IVR **YES**
TRDN NONE
ALOG **YES**
OBSC NO
OBPT 5
CWNT NONE

b. Create CDN by overlay 23. Below is output of CDN 3111 after created.

>ld 23
ACD000
MEM AVAIL: (U/P): 103093188   USED U P: 481334 77827   TOT: 103652349
DISK SPACE NEEDED: 60 KBYTES
ACD DNS   AVAIL: 23997   USED: 3   TOT: 24000
REQ prt
MEM AVAIL: (U/P): 103093188   USED U P: 481334 77827   TOT: 103652349
DISK SPACE NEEDED: 60 KBYTES

ACD DNS AVAIL: 23997 USED: 3 TOT: 24000
REQ prt
TYPE **cdn**
CUST 0
CDN **3111**

TYPE **CDN**
CUST 0
CDN **3111**
FRRT
SRRT
FROA NO
UUI NO
MURT
CDSQ NO
DFDN **3109** <--- default forward to ACD
NAME NO
CMB NO
CEIL 2047
OVFL NO
TDNS NO
----

c. Create some 2008 agents by ld 11. Below is configuration output of one agent after created.

>ld 11
SL1000
MEM AVAIL: (U/P): 103093188 USED U P: 481334 77827 TOT: 103652349
DISK SPACE NEEDED: 60 KBYTES
TNS AVAIL: 32597 USED: 170 TOT: 32767

REQ: prt
TYPE: **2008**

TN **4 0 9 0**
DATE
PAGE
DES

DES 2008
TN 004 0 09 00 VIRTUAL
TYPE 2008
CDEN 8D
CTYP XDLC
CUST 0
ERL 0
FDN
TGAR **1**
LDN NO
NCOS 0
SGRP 0

RNPG 0
SCI 0
SSU
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS **UNR** FBD WTA LPR MTD FND HTD ADD HFD
MWD LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD DSX VMD MMA SLKD CCSD SWD LND CNDD
CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
ICDD CDMD LLCN MCTD CLBD AUTU
GPUD DPUD DNDD CFXD ARHD CNTD CLTD ASCD
CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
UDI RCC HBTD AHD IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
DRDD EXR0
USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3
MCBN
FDSD NOVD CDMR PRED MCDD T87D PKCH
CPND_LANG ENG
HUNT
PLEV 02
PUID
DANI NO
SPID NONE
AST
IAPG 0
AACS YES
ACQ AS: TN
ASID 16
SFNB 1 2 3 5 6 9 10 11 12 13 15 16 17 18 19 21 22 23 32 33 34 35 36 37 38
SFRB 32 33 34 35 36 37 38
USFB 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15
CALB 0 1 2 3 4 5 6 7 8 9 10 11 12
FCTB
ITNA NO
DGRP
PRI 01
MLWU_LANG 0
DNDR 0
KEY **00 ACD 3109 0 3201**
AGN
01 SCN 3301 0 MARP
02
03
04
05
06
07
DATE 12 NOV 2009
NACT

d. Create ELAN – VAS by ld 17. Below are configuration outputs of them after created.

```
>ld 22
PT2000

REQ  prt
TYPE adan elan 16

ADAN     ELAN 16
  CTYP ELAN
  DES  CPilot
  N1   512

REQ

>ld 22
PT2000

REQ  prt
TYPE vas

VAS
  VSID   016
    DLOP
    ELAN 016
      SECU YES
      INTL 0001
      MCNT 9999
```

e. Check the link between CS1000 and Callpilot

```
>ld 48
LNK000
.stat elan

SERVER TASK:  ENABLED
ELAN #: 016  DES: CPilot
   APPL_IP_ID: 47 .248 .100 .151 LYR7: ACTIVE  EMPTY   APPL ACTIVE
```

4.9. CS1000 SIP-Line Configuration

In this section, it shows how to configure a SIP LINE system on CS1000. Follow the below steps to setup the SIP LINE server.

4.9.1. Configure SIP LINE CS1000 in Element Manager

This section shows how to configure SIP LINE Node 1002. For adding a new node IP telephony, please refer to Section 4.2.

a) Log in UCM and EM (please refer to Section 4.1.1)

b) Under **System** -> **IP Network** -> **IP Telephony Nodes.**

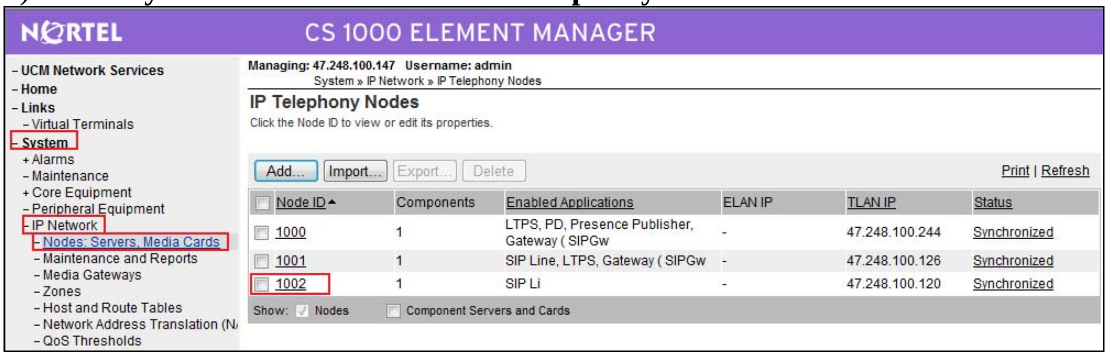

Figure 69 – Add a new node for SIP line.

- Enter the **host IP Address, ELAN Gateway IP Address** and then click on **SIP Line**.

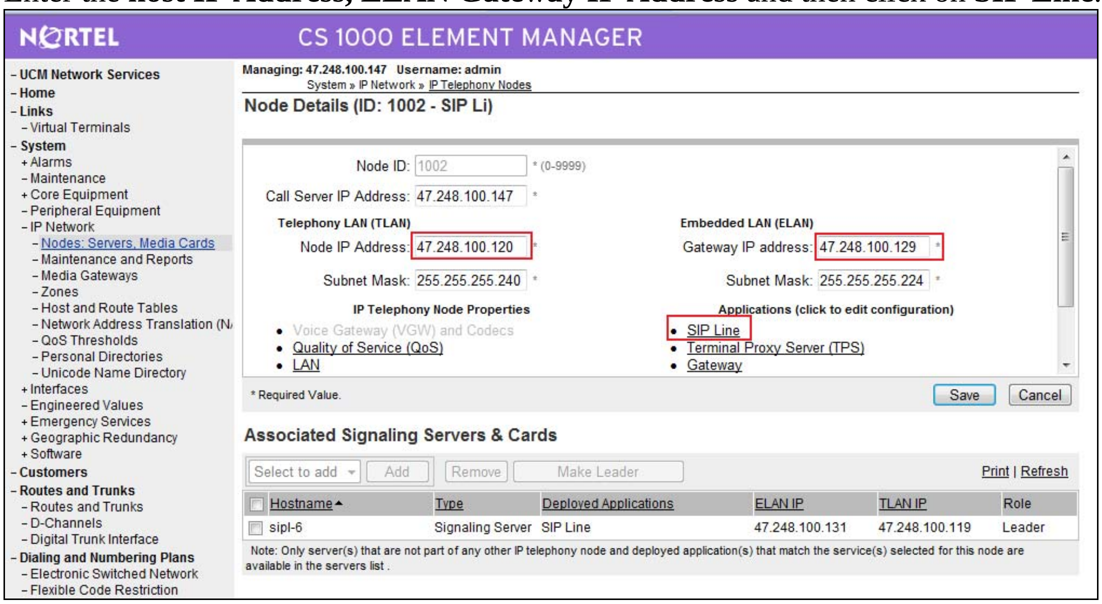

Figure 70 –Configure SIP line.

- Enter **SIP Domain Name** (check this on SPS) and **SLG Group ID** (this is Node ID)

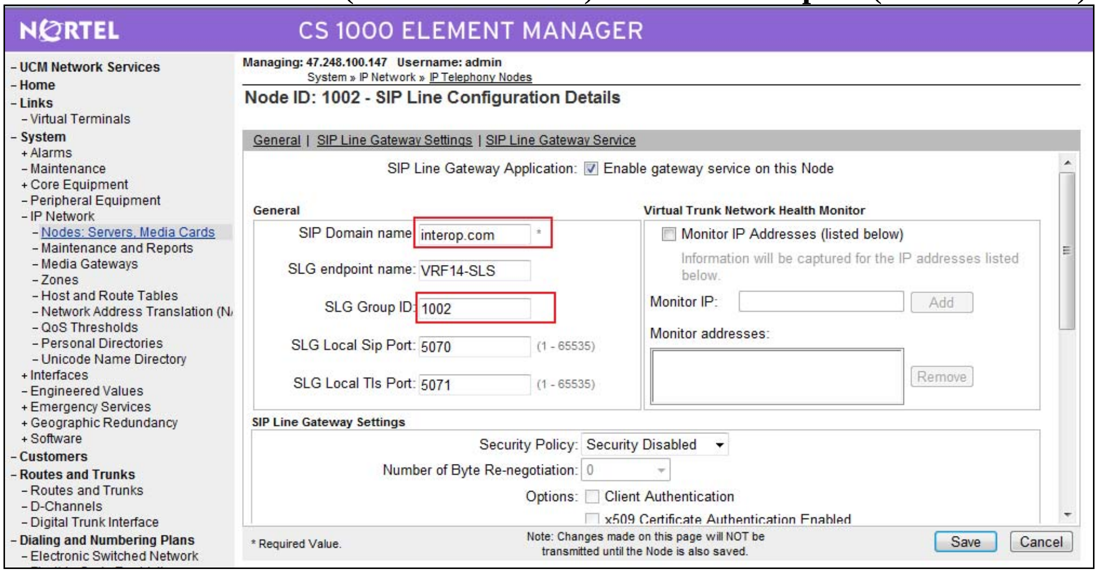

Figure 71 – SIP-Line Configuration Details Page.

- Select **SLG Role**. enter **MO SLG IP** and **MO SLG Port** as shown in **Figure 72.**

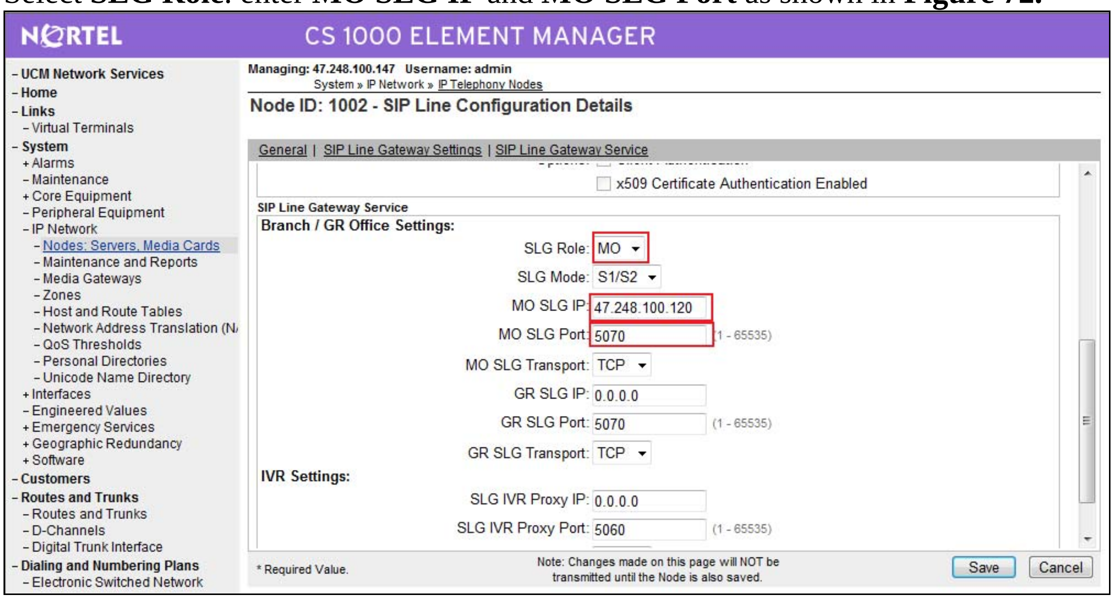

Figure 72 – SIP-Line Configuration Details Page.

- Click **Save** and then **SYNC** is required. Please refer to Section 4.2.4 for more detail.

4.9.2. Packages Required for SIP line on CS1000 Call Server

1. SLS_Package – 417 - SIP Line Service
2. FFC- 139 - Flexible Feature Codes
3. SIP_LINE_NT_PKG – 415 - Nortel SIP Line Package

4. SIP_LINE_3P_PKG – 416 - 3rdParty SIP Line Package

4.9.3. Configure SIPL service in LD15

LD 15
REQ CHG
TYPE SLS
CUST 0
SIPL_ON **YES**
SIPD **INTEROP.COM**
UAPR **222** - DN prefix used to auto-generate UADN for all SIPL clients of this customer
NMME NO

4.9.4. Configure DCH for SIPL in LD 17

```
LD 17
REQ CHG
TYPE ADAN
ADAN new dch 11
ADAN     DCH 11
  CTYP DCIP
  DES  SIPL
  USR  ISLD
  ISLM 4000
  SSRC 1800
  OTBF 32
  NASA NO
  IFC  SL1
  CNEG 1
  RLS  ID  25
  RCAP
  MBGA NO
  H323
    OVLR NO
    OVLS NO
```

4.9.5. Configure ELAN AML link in LD 17

LD 17
REQ CHG
TYPE ADAN
ADAN new elan 32
ADAN   ELAN **32 –** new AML ELAN link, link number should be bigger or equal to 32
  CTYP **ELAN**
  DES **SIPL**
  N1   512

4.9.6. Configure VAS ID for AML link in LD 17

LD 17
REQ CHG
TYPE VAS
VAS new
VSID **32** – VAS ID number
ELAN **32** – Defined in step 3

4.9.7. Configure SIPL route

LD 16
REQ new
TYPE rdb
CUST 0
ROUTE 11
DES **SIPL**
TKTP **TIE**
…
VTRK **YES**
ZONE **10 –** virtual trunk zone defined in LD117
PCID **SIPL**
…
NODE **1002 –** node ID of SIPL node
DTRK NO
ISDN YES
MODE **ISLD**
DCH **11** – DCH defined in step 2
IFC **SL1**
PNI **00001**
NCNA **YES**
NCRD **YES**
TRO NO
FALT NO
CTYP UKWN
INAC **YES**
ISAR NO
DAPC NO
…
ICOG **IAO**
…
ACOD **8011 –** route access code

4.9.8. Configure SIPL trunks

LD 14
REQ **NEW 256 –** e.g. create 256 trunks
TYPE **IPTI**
TN **124 0 0 0** - starting TN for virtual trunks

DES **SIPL**
CUST 0
RTMB **11 1** – route number and member
CHID 1
TGAR **0**
STRI **IMM**
STRO **IMM**
CLS **UNR**

4.9.9. Check status of SIP-Line link and SIP line Gateway

On Call Server

>*ld 96
DCH 011 : OPER EST ACTV AUTO DES : SIPL_N1402

On SLG

[nortel@vrf14-sls ~]$ slgShow
=== VTRK ===

===== General ======
SLG State = AppReady
Total User Registered = 1

===== AML Info ======

hAppBlk	TaskName	Tid	LinkState	NumRetry	LinkNum	Trace
0x1226c80	SLG	0xfb00	**Up**	0	32	0

4.9.10. Setting password length for SIP line

LD 15
REQ CHG
TYPE: **FFC**
TYPE FFC_DATA
CUST 0

SCPL **4 –** password length is 4

4.9.11. Provisioning SIP client accounts on CS1000 Call Server

LD 11
REQ NEW
TYPE UEXT

TN **104 0 00 11** - Virtual TN for SIPL client
CUST **0**
UXTY **SIPL** – UEXT type must be SIPL
MCCL **YES**

SIPN 1
SIP3 1
FMCL 0
TLSV 0

** Begin Note:
For SIP Nortel phones: SIPN-SIP3-FMCL-TLSV = 1-0-0-0
For SMC3456: SIPN-SIP3-FMCL-TLSV = 1-0-0-0
For 3party SIP phones: SIPN-SIP3-FMCL-TLSV = 0-1-0-0

***End Note

SIPU **4197** – SIPL userID, often set equal to DN of the phone
NDID **1002 –** NodeID of the SIPL node
ZONE 001 – MO zone configured in LD 117
TGAR **0**
…
SCPW **1234 –** password for SIPL client to log in
…
CLS **UNR**
…
KEY 00 SCR **4197** – DN of the phone
    CPND **NEW** – in case you want to set CLID for phone
        NAME **set4197**
        XPLN 20
        DISPLAY_FMT FIRST,LAST
  01 HOT U **2224197**

4.9.12. Check current status of set registration on SLG

```
[nortel@vrf14-sls ~]$ slgSetShowAll

=== VTRK ===
UserID          TN              Clients  Calls  SetHandle
--------------- --------------- -------  -----  ---------
           4861  104-00-00-11         1      0  0xb7d8a0c8
```

4.9.13. SMC3456 Softphone Installation

Link to download: http://livelink-ott.ca.nortel.com/livelink/livelink.exe?func=ll&objId=34471954&objAction=browse&sort=name&viewType=1.

a) After installation on the PC and apply the Licence key which is required for activate the SMC to be used. Run the SMC3456, you will see the **Figure 73**. Enter any username and password then click **Sign in.**

Figure 73 – SMC3456 Log In.

b) Click **Skip**

Figure 74 – SMC3456 Log In (Con.)

c) The SMC3456 Client will be displayed as follows.

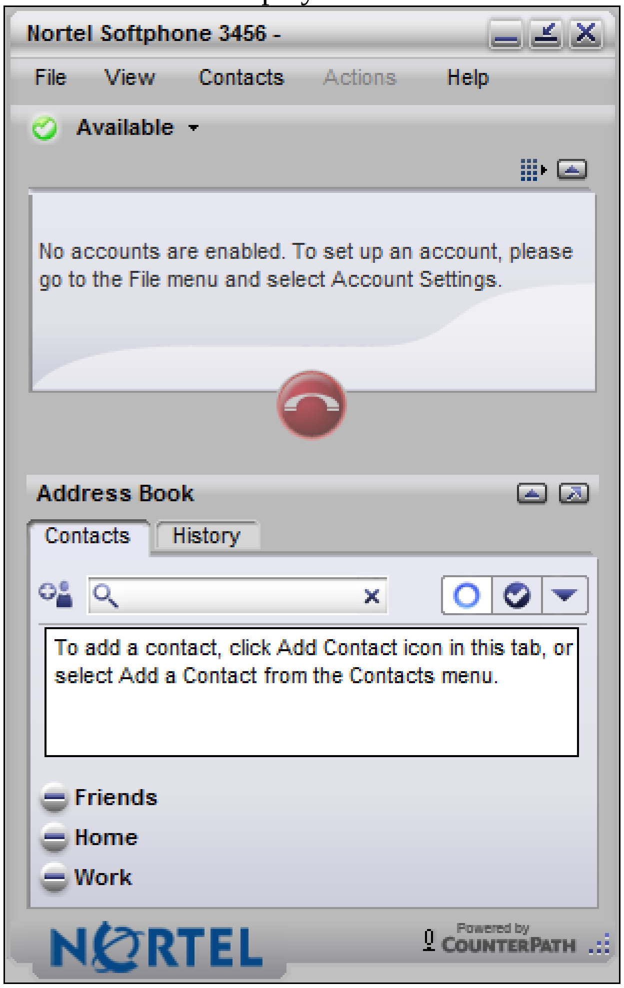

Figure 75 - SMC3456 Client

d) On the top menu bar, go to **FILE** -> **Preferences. Preferences** screen will display. Click **Advanced**. At **Log in Server** tab, check **No log in server available** as shown in **Figure 76**.

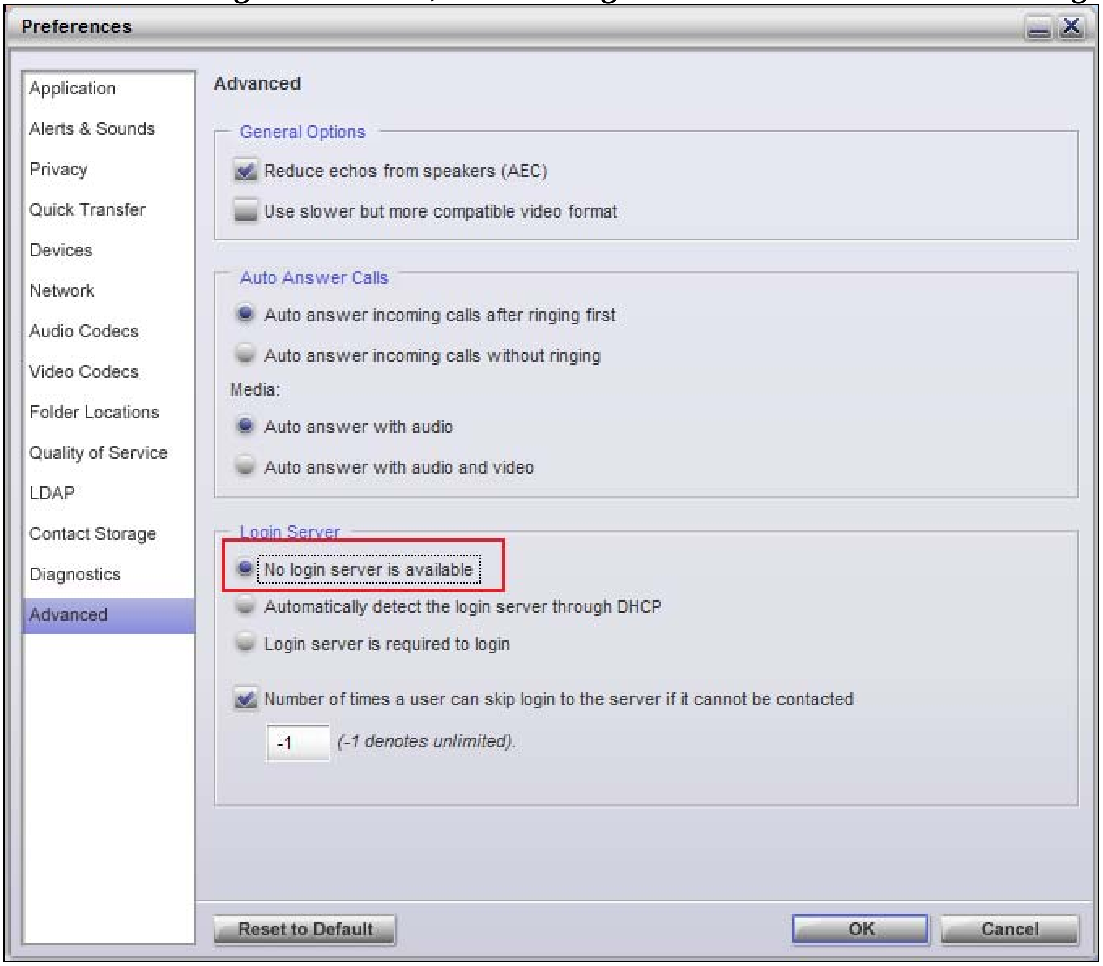

Figure 76 – Advanced Options Menu

4.9.14. Add a SIP Account on SMC3456

a) In order to create a SIP account for SMC3456 to be able to register to CS1000 SIP line server, from the top menu bar, go to **FILE** -> **ACCOUNT SETTINGS** and then click **Edit**, please see **Figure 77**.

Figure 77 – Accounting Settings

b) The created account is appeared as **Figure 78**.

Figure 78 – SIP Account Detail Settings

c) **Figure 79** shows the newly created SIP account.

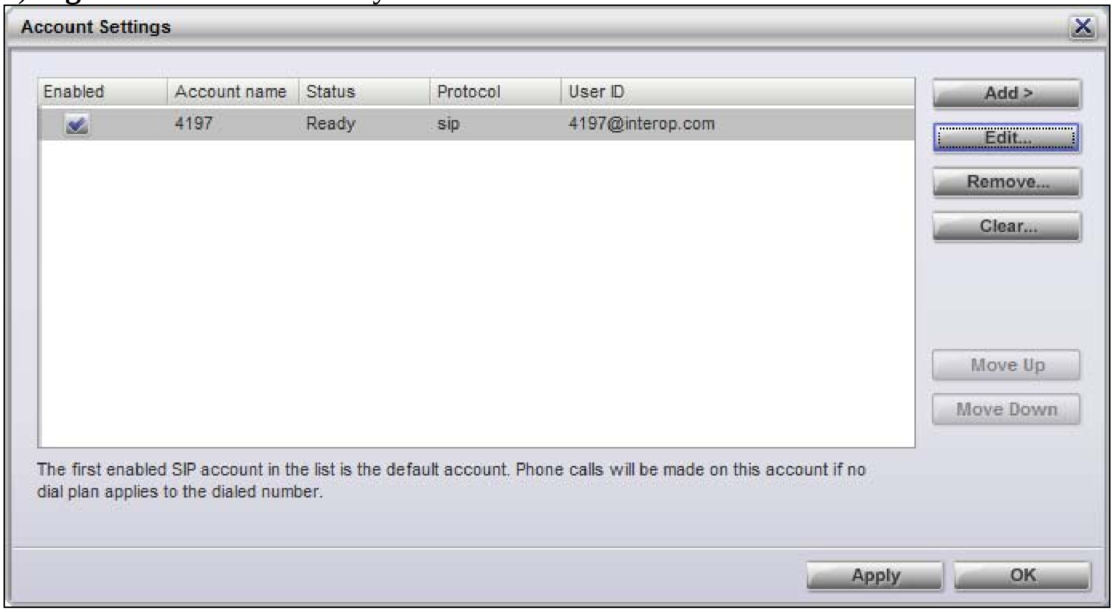

Figure 79 – New Created SIP Account

4.10. CS1000 Tandem Configuration

In this section, this application note assumes that the basic configuration has already been administered. The below procedures describe the configuration details of Avaya Communication Server 1000 A, Communication Server 1000 B and SPS. For further information on Avaya Communications Server 1000, please consult reference in Section 9.

4.10.1. Network topology for multi-system (tandem calls)

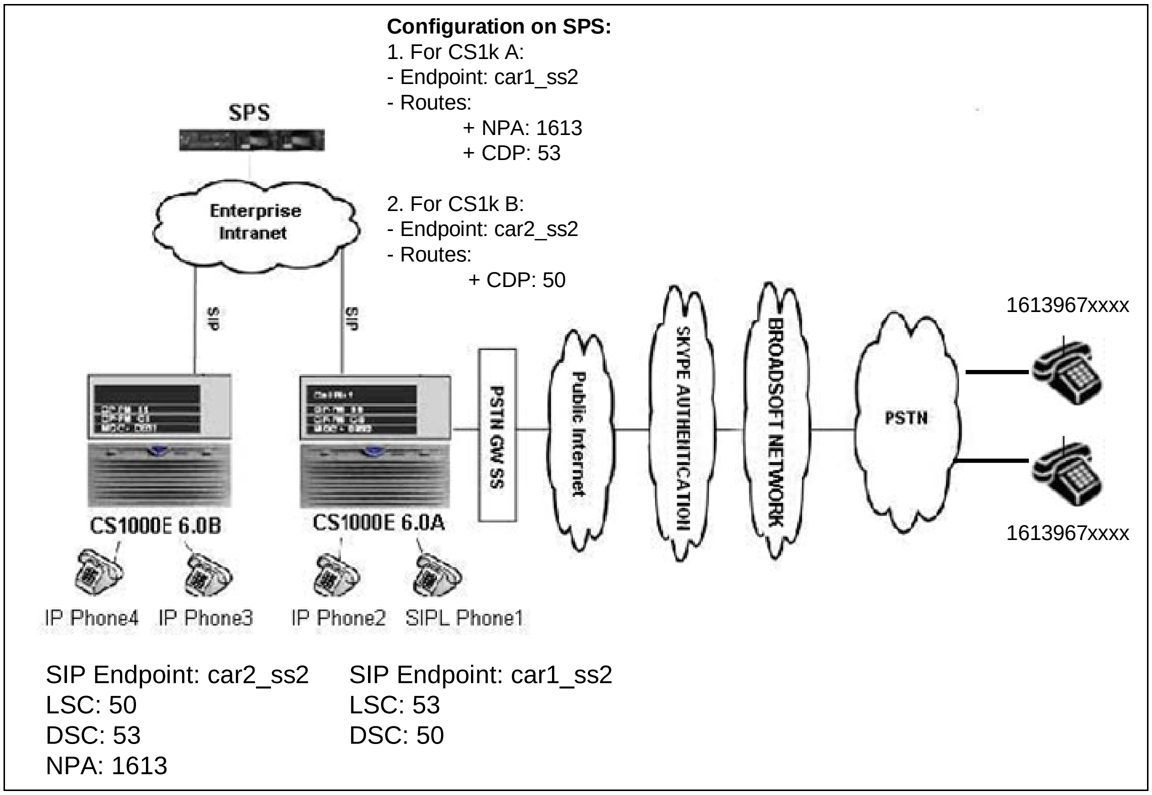

Figure 80 – Network Topology for Multi-system (tandem calls)

4.10.2. Avaya Communication Server 1000 A

4.10.2.1 Configure or add new node IP Telephony

To configure or add new node IP telephony to CS1000 A, please follow Section 4.2. For this new node, user needs to add more hardware such as CPPM, COT....etc.

4.10.2.2 Configure SIP Trunk Gateway

To configure a SIP Trunk to SPS, please follow Section 4.5 for more detail. The difference is this trunk is configured to register to SPS instead of Skype.

In this section, this application note just shows the configuration details of configured SIP trunk.

a. Administer Virtual D-Channel

```
>ld 22
PT2000

REQ  prt
TYPE adan dch 101

ADAN     DCH 101
  CTYP DCIP
  DES  Enterprise
  USR  ISLD
  ISLM 4000
  SSRC 1800
  OTBF 32
  NASA YES
  IFC  SL1
  CNEG 1
  RLS  ID  25
  RCAP ND2 MWI
  MBGA NO
  H323
    OVLR NO
    OVLS NO

REQ
```

b. Administer Virtual SIP Routes

```
>ld 21
PT1000

REQ: prt
TYPE: rdb
CUST 0
ROUT 101

TYPE RDB
CUST 00
ROUT 101
DES  ENTERPRISE
TKTP TIE
M911P NO
```

```
ESN  YES
RPA  NO
CNVT NO
SAT  NO
IDEF NET
RCLS EXT
VTRK YES
ZONE 255
PCID SIP
CRID NO
NODE 1001
DTRK NO
ISDN YES
   MODE ISLD
   DCH  101
   IFC  SL1
   PNI  00001
   NCNA YES
   NCRD YES
   TRO  NO
   FALT NO
   CTYP NPA
   INAC YES
   ISAR NO
   DAPC NO
PTYP ATT
AUTO NO
DNIS NO
DCDR NO
ICOG IAO
SRCH LIN
TRMB YES
STEP
ACOD 8101
TCPP NO
PII NO
AUXP NO
TARG 01
CLEN 10
BILN NO
OABS
INST
IDC  YES
DCNO 1
NDNO 1 *
DEXT NO
DNAM NO
ANTK
SIGO STD
STYP SDAT
MFC  NO
ICIS YES
```

```
OGIS YES
PTUT 0
TIMR ICF  512
    OGF  512
    EOD  13952


PAGE 002

    DSI  34944
    NRD  10112
    DDL  70
    ODT  4096
    RGV  640
    GTO  896
    GTI  896
    SFB  3
    NBS  2048
    NBL  4096

    IENB  5
    TFD  0
    VSS  0
    VGD  6
    EESD  1024
SST  5 0
DTD  NO
SCDT NO
2 DT NO
NEDC ORG
FEDC ORG
CPDC NO
DLTN NO
HOLD 02 02 40
SEIZ 02 02
SVFL 02 02
DRNG NO
CDR  NO
NATL YES
SSL
CFWR NO
IDOP NO
VRAT NO
MUS  NO
PANS YES
MANO NO
FRL  0 0
FRL  1 0
FRL  2 0
FRL  3 0
FRL  4 0
FRL  5 0
```

```
FRL  6 0
FRL  7 0
OHQ  NO
OHQT 00
CBQ  NO
AUTH NO
TDET NO
TTBL 0
ATAN NO
OHTD NO
PLEV 2
OPR  NO
ALRM NO
ART  0
PECL NO
DCTI 0
TIDY 8101 101
ATRR NO
TRRL NO


PAGE 003

SGRP 0
CCBA NO
ARDN NO
AACR NO

REQ
```

c. Administer Virtual Trunks

```
>ld 20

PT0000
REQ: prt
TYPE: ipti
TN   100 0 1 21
DATE
PAGE

DES  ENTER
TN   100 0 01 21  VIRTUAL
TYPE IPTI
CDEN 8D
CUST 0
XTRK VTRK
ZONE 255
LDOP BOP
TIMP 600
BIMP 600
AUTO_BIMP NO
```

NMUS NO
TRK ANLG
NCOS 7
RTMB 101 22
CHID 22
TGAR 0
STRI/STRO IMM IMM
SUPN YES
AST NO
IAPG 0
CLS UNR DTN CND ECD WTA LPR APN THFD XREP SPCD MSNV
P10 NTC MID
TKID
AACR NO
DATE 27 AUG 2010

NACT

d. Administer SIP trunk gateway to SPS.

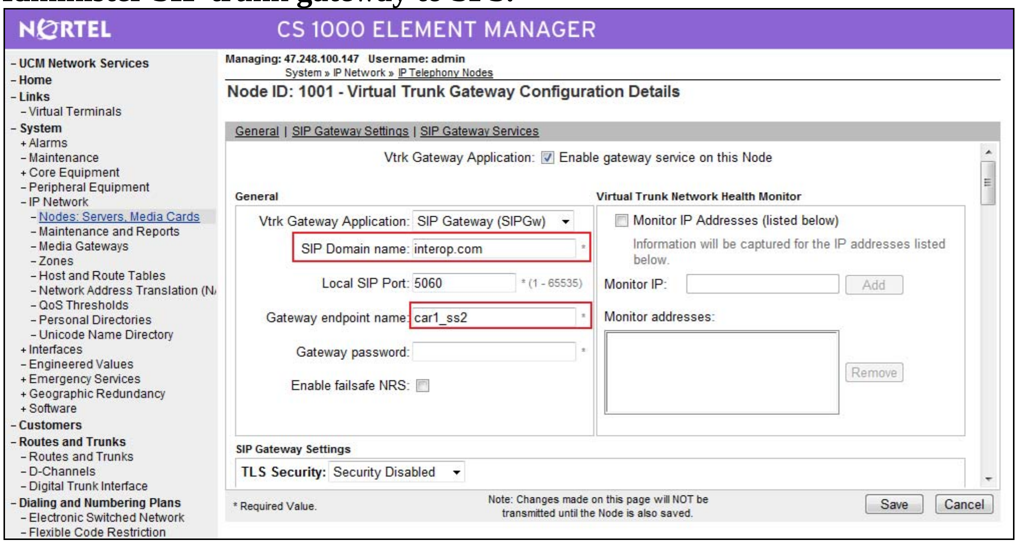

Figure 81 – Virtual trunk gateway Configuration Details.

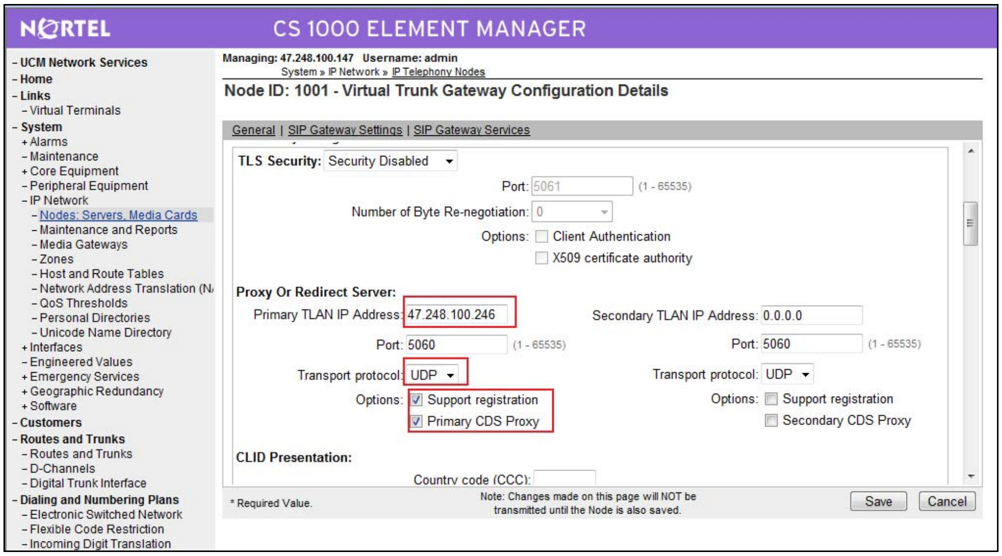

Figure 82 – Virtual Trunk Gateway Configuration Details.

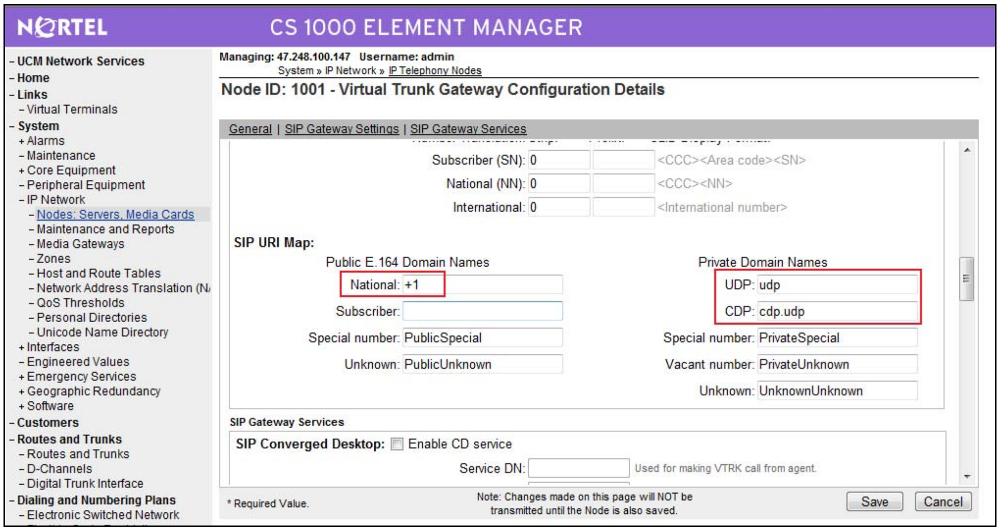

Figure 83 – Virtual trunk gateway Configuration Details.

4.10.2.3 Coordinated Dialing Plan (CDP) - outbound call to CS1000_B

a) Create Digit Manipulation Block 50 as shown in **Figure 84**

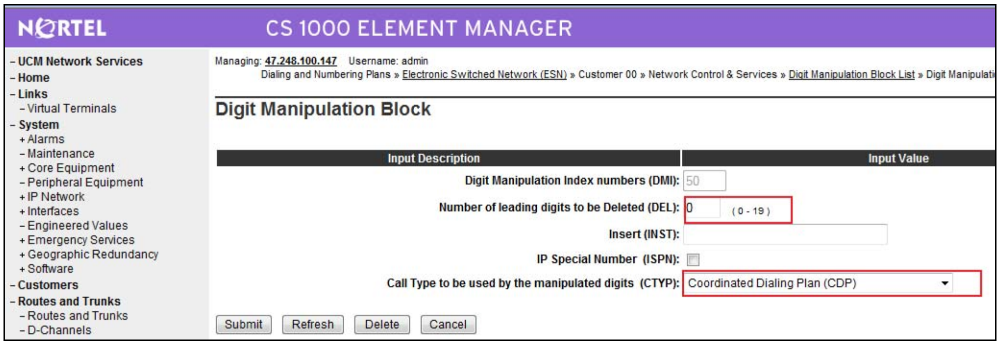

Figure 84 – DMI Configuration Details.

b) Create Route List Blocks 50 as shown in **Figure 85**.

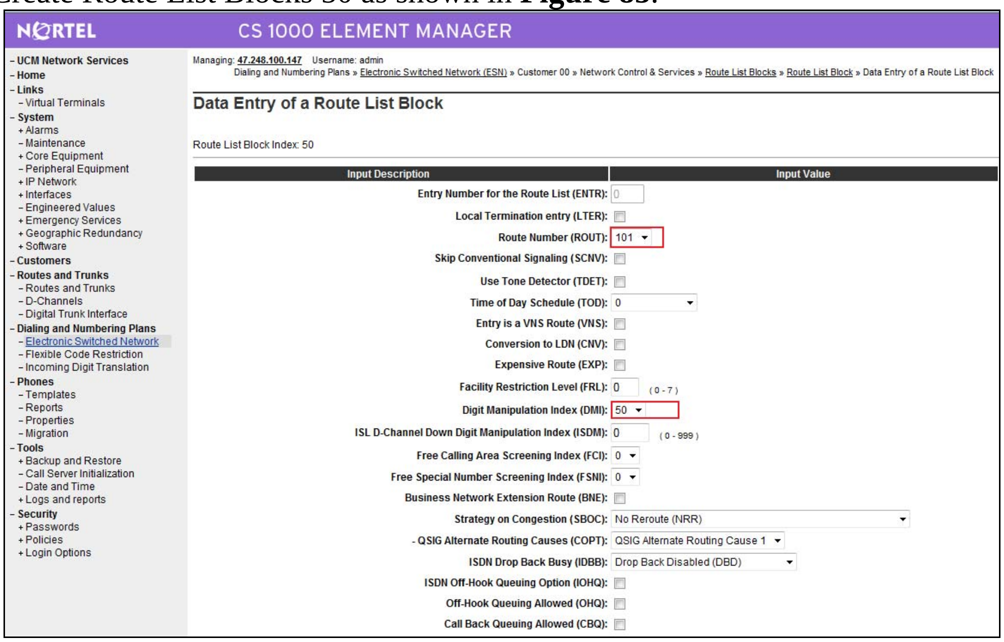

Figure 85 – Route List Blocks Configuration Details.

c) Create Distant Steering Code (DSC) to route the call to CS1000_B.

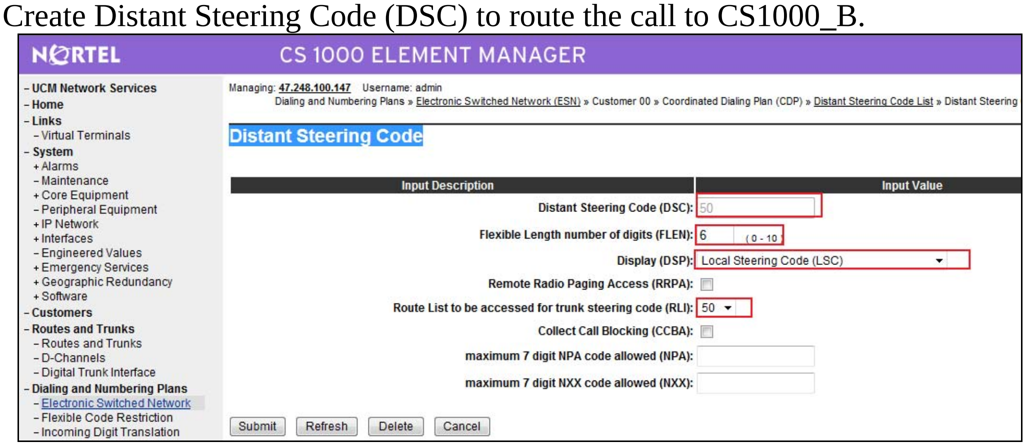

Figure 86 – Distant Steering Code Configuration Details.

4.10.2.4 Coordinated Dialing Plan (CDP) - Inbound call

a) Create Digit Manipulation Block 2 as shown in **Figure 87.**

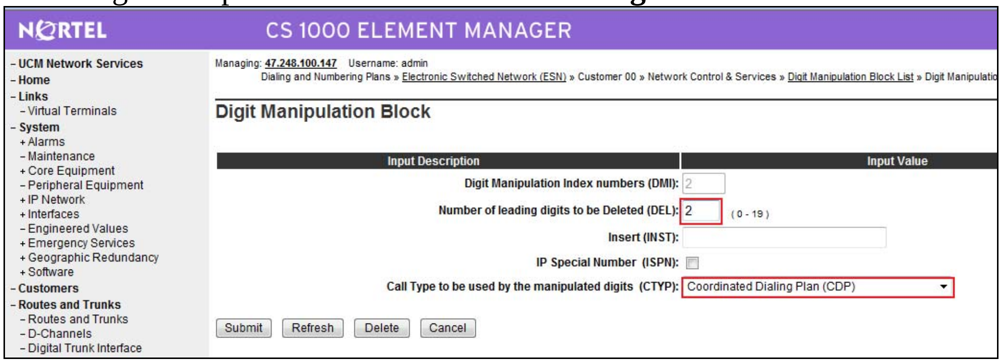

Figure 87 – DMI Configuration Details.

b) Create Local Steering Code (LSC) to receive the calls.

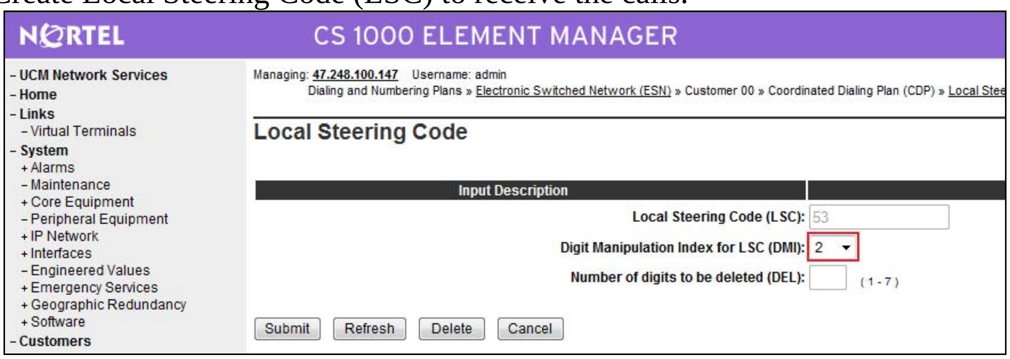

Figure 88 – Local Steering Code Configuration Details.

4.10.2.5 Configure Dialing Plan - route a call from PSTN to CS1000_B

When there is a call from PSTN to the online number 12107574598, this call will come to CS1000_A first. If user wants to receive this call by a phone on CS1000_B, user has to configure on CS1000_A to forward it to CS1000_B. To re-route a call from CS1000_A to CS1000_B, user can use IDC feature as follows.

a) Configure FCR in Customer by ld 15. This section prints FCR configuration details.

```
>ld 21
PT1000

REQ: prt
TYPE: fcr
```

```
TYPE FCR_DATA
CUST 0

TYPE FCR_DATA
CUST 00
NFCR YES
  MAXT 100
  OCB1 255
  OCB2 255
  OCB3 255
IDCA YES
  DCMX 100
```

b) Configure IDC by ld 49. This section prints IDC configuration details.

```
>ld 49
DGT000
MEM AVAIL: (U/P): 103093188    USED U P: 481334 77827    TOT: 103652349
DISK SPACE NEEDED: 60 KBYTES
REQ  prt
TYPE idc
CUST 0
DCNO

DCNO    1 <----- this number is configured in Rout 100,
              ******** Note *********
                     LD 16
                           ROUT 100
                           ...................
                           IDC  YES
                           DCNO 1
                           NDNO 1 *
              *********End Note *********
SDID  NO
IDGT    CDGT
12107574698    504698
13157914457    3111

MEM AVAIL: (U/P): 103093188    USED U P: 481334 77827    TOT: 103652349
DISK SPACE NEEDED: 60 KBYTES
REQ
```

User is also able to configure IDC via UCM-EM as shown in **Figure 89**.

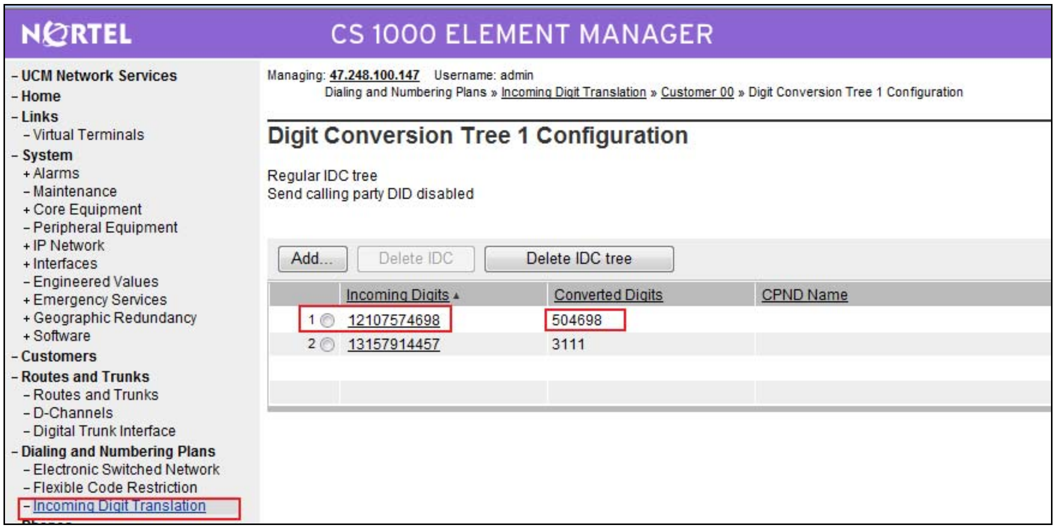

Figure 89 – Digit Conversion Tree 1 Configuration Details.

4.10.3. Avaya Communication Server 1000 B

4.10.3.1 Configure or add a new node IP Telephony

Please follow Section 4.2 for more detail.

4.10.3.2 Configure SIP Trunk Gateway

To configure a SIP Trunk to SPS, please follow Section 4.5 for more detail. The difference is this trunk is configured to register to SPS instead of Skype.

In this section, this application note just shows the configuration details of configured SIP trunk.

a. Administer Virtual D-Channel

```
>ld 22
PT2000

REQ  prt
TYPE adan dch 101

ADAN     DCH 101
  CTYP DCIP
  DES  Enterprise
  USR  ISLD
  ISLM 4000
  SSRC 1800
  OTBF 32
  NASA YES
```

```
  IFC  SL1
  CNEG 1
  RLS  ID  25
  RCAP ND2 MWI
  MBGA NO
  H323
    OVLR NO
    OVLS NO

REQ
```

b. Administer Virtual SIP Routes

```
>ld 21
PT1000

REQ: prt
TYPE: rdb
CUST 0
ROUT 101

TYPE RDB
CUST 00
ROUT 101
DES  ENTERPRISE
TKTP TIE
M911P NO
ESN  NO
RPA  NO
CNVT NO
SAT  NO
IDEF NET
RCLS EXT
VTRK YES
ZONE 255
PCID SIP
CRID NO
NODE 2001
DTRK NO
ISDN YES
    MODE ISLD
    DCH  101
    IFC  SL1
    PNI  00001
    NCNA YES
    NCRD YES
    TRO  NO
    FALT NO
    CTYP UKWN
    INAC YES
    ISAR NO
    DAPC NO
```

```
MBXR NO
PTYP ATT
AUTO NO
DNIS NO
DCDR NO
ICOG IAO
SRCH LIN
TRMB YES
STEP
ACOD 8101
TCPP NO
PII NO
AUXP NO
TARG 01
CLEN 10
BILN NO
OABS
INST
IDC  YES
DCNO 1
NDNO 1 *
DEXT NO
DNAM NO
ANTK
SIGO STD
STYP SDAT
MFC  NO
ICIS YES
OGIS YES
PTUT 0
TIMR ICF  512
     OGF  512


PAGE 002

     EOD  13952
     DSI  34944
     NRD  10112
     DDL  70
     ODT  4096
     RGV  640
     GTO  896
     GTI  896
     SFB  3
     NBS  2048
     NBL  4096

     IENB  5
     TFD  0
     VSS  0
     VGD  6
```

EESD 1024
SST 5 0
DTD NO
SCDT NO
2 DT NO
NEDC ORG
FEDC ORG
CPDC NO
DLTN NO
HOLD 02 02 40
SEIZ 02 02
SVFL 02 02
DRNG NO
CDR NO
NATL YES
SSL
CFWR NO
IDOP NO
VRAT NO
MUS YES
MRT 50
PANS YES
MANO NO
FRL 0 0
FRL 1 0
FRL 2 0
FRL 3 0
FRL 4 0
FRL 5 0
FRL 6 0
FRL 7 0
OHQ NO
OHQT 00
CBQ NO
AUTH NO
TDET NO
TTBL 0
ATAN NO
OHTD NO
PLEV 2
OPR NO
ALRM NO
ART 0
PECL NO
DCTI 0
TIDY 8101 101

PAGE 003

ATRR NO
TRRL NO

```
SGRP 0
CCBA NO
ARDN NO
AACR NO

REQ:
```

c. Administer Virtual Trunks

```
>ld 20

PT0000
REQ: prt
TYPE: ipti
TN   100 0 1 0
DATE
PAGE

DES  ENTER
TN   100 0 01 00  VIRTUAL
TYPE IPTI
CDEN 8D
CUST 0
XTRK VTRK
ZONE 255
LDOP BOP
TIMP 600
BIMP 600
AUTO_BIMP NO
NMUS NO
TRK  ANLG
NCOS 7
RTMB 101 1
CHID 1
TGAR 0
STRI/STRO IMM IMM
SUPN YES
AST  NO
IAPG 0
CLS  UNR DTN CND ECD WTA LPR APN THFD XREP SPCD MSNV
    P10 NTC MID
TKID
AACR NO
DATE 27 AUG 2010


NACT
```

d. Administer SIP trunk gateway to SPS.

Basing on network topology shown in Section 4.10.1, below is the configuration details of SIP trunk on CS1000_B to SPS.

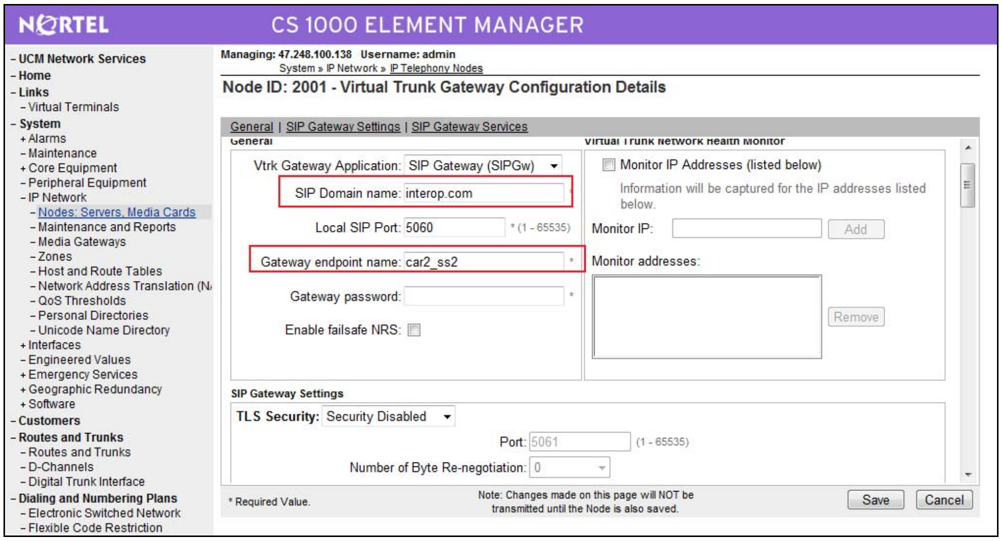

Figure 90 – Virtual Trunk Gateway Configuration Details.

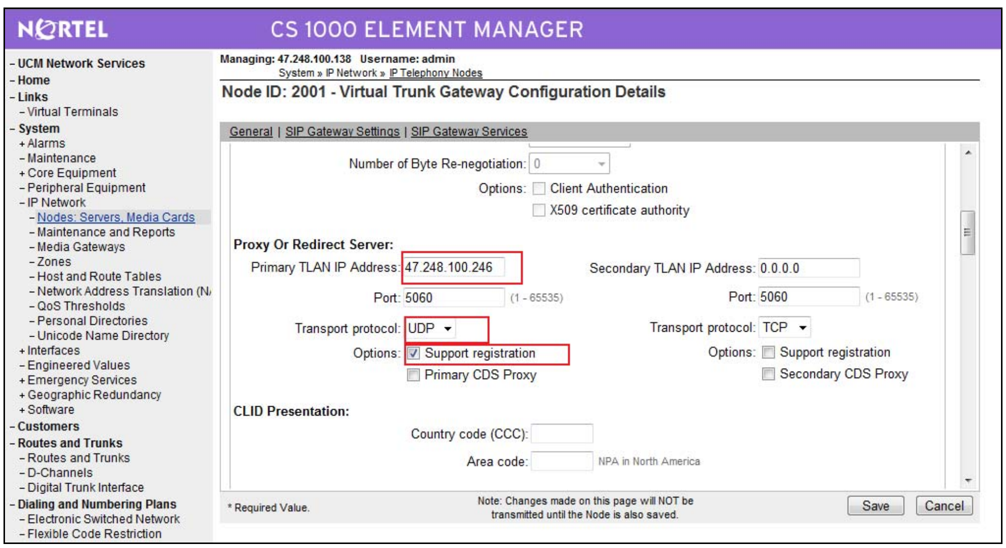

Figure 91 – Virtual trunk gateway configuration details

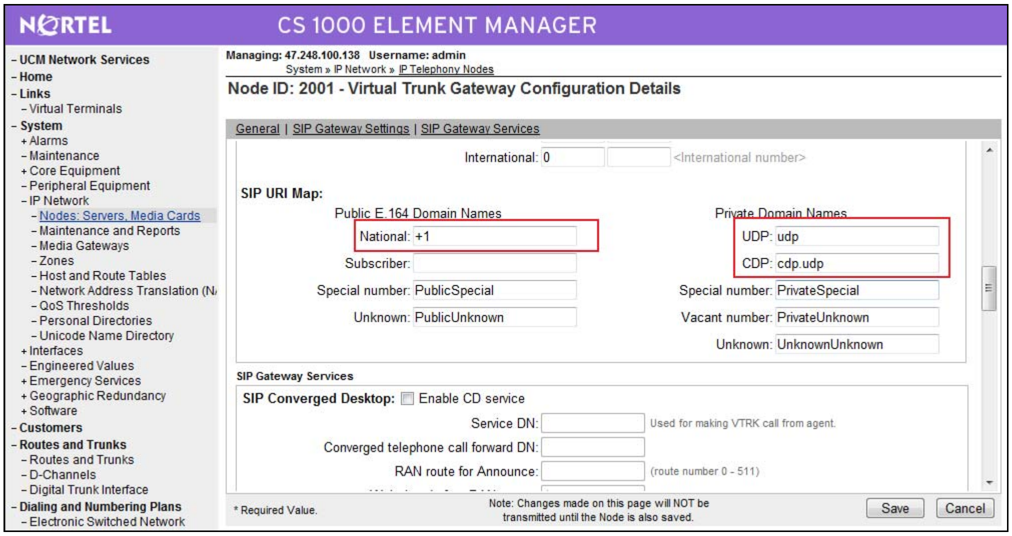

Figure 92 – Virtual Trunk Gateway Configuration Details.

4.10.3.3 Coordinated Dialing Plan (CDP) - Outbound call to CS1000_A

a) Create Digit Manipulation Block 53

>ld 86
ESN000

MEM AVAIL: (U/P): 37549072 USED U P: 523769 95732 TOT: 38168573
DISK SPACE NEEDED: 97 KBYTES
REQ prt
CUST 0
FEAT dgt
DMI 53

DMI 53
DEL 0
ISPN NO
CTYP **CDP**

b) Create Route List Blocks 53

>ld 86
ESN000

MEM AVAIL: (U/P): 37549072 USED U P: 523769 95732 TOT: 38168573
DISK SPACE NEEDED: 97 KBYTES
REQ ptr

ESN004
REQ prt
CUST 0
FEAT rlb
RLI 53

RLI 53
ENTR 0
LTER NO
ROUT 101
TOD 0 ON 1 ON 2 ON 3 ON
 4 ON 5 ON 6 ON 7 ON
VNS NO
SCNV NO
CNV NO
EXP NO
FRL 0
DMI 53
ISDM 0
FCI 0
FSNI 0
BNE NO
DORG NO
SBOC NRR
IDBB DBD
IOHQ NO
OHQ NO
CBQ NO

ISET 0
NALT 5
MFRL 0
OVLL 0

MEM AVAIL: (U/P): 37549072 USED U P: 523769 95732 TOT: 38168573
DISK SPACE NEEDED: 97 KBYTES

c) Create Distant Steering Code (DSC) to route the call to CS1000_B.

>ld 87

MEM AVAIL: (U/P): 37549072 USED U P: 523769 95732 TOT: 38168573
DISK SPACE NEEDED: 97 KBYTES
REQ prt
CUST 0
FEAT cdp
TYPE dsc
DSC 53
DSC 53
FLEN 6
DSP LSC
RRPA NO
RLI 53
CCBA NO
NPA
NXX

```
MEM AVAIL: (U/P): 37549072    USED U P: 523769 95732    TOT: 38168573
DISK SPACE NEEDED: 97 KBYTES
```

4.10.3.4 Coordinated Dialing Plan (CDP) - Inbound call

a) Create Digit Manipulation Block 2

```
>ld 86
ESN000

MEM AVAIL: (U/P): 37549072    USED U P: 523769 95732    TOT: 38168573
DISK SPACE NEEDED: 97 KBYTES
REQ  prt
CUST 0
FEAT dgt
DMI  2


DMI  2
DEL  2
ISPN NO
CTYP CDP

MEM AVAIL: (U/P): 37549072    USED U P: 523769 95732    TOT: 38168573
DISK SPACE NEEDED: 97 KBYTES
```

b) Create Local Steering Code (LSC) to receive the calls.

```
>ld 87
ESN000

MEM AVAIL: (U/P): 37549072    USED U P: 523769 95732    TOT: 38168573
DISK SPACE NEEDED: 97 KBYTES
REQ  prt
CUST 0
FEAT cdp
TYPE lsc
LSC  50
LSC  50
DMI  2

MEM AVAIL: (U/P): 37549072    USED U P: 523769 95732    TOT: 38168573
DISK SPACE NEEDED: 97 KBYTES
```

4.10.3.5 Configure Dialing Plan – Outbound call to PSTN via CS1000_A

a) Create Digit Manipulation Block – DMI 10

```
>ld 86
ESN000
```

```
MEM AVAIL: (U/P): 37549072    USED U P: 523769 95732    TOT: 38168573
DISK SPACE NEEDED: 97 KBYTES
REQ  prt
CUST 0
FEAT dgt
DMI  10


DMI  10
DEL  0
ISPN NO
CTYP NPA


MEM AVAIL: (U/P): 37549072    USED U P: 523769 95732    TOT: 38168573
DISK SPACE NEEDED: 97 KBYTES
```

b) Create Route List Blocks – RLB 10

```
>ld 86
ESN000

MEM AVAIL: (U/P): 37549072    USED U P: 523769 95732    TOT: 38168573
DISK SPACE NEEDED: 97 KBYTES
REQ  prt
CUST 0
FEAT rlb
RLI  10

RLI  10
ENTR 0
LTER NO
ROUT 101
TOD  0 ON  1 ON  2 ON  3 ON
     4 ON  5 ON  6 ON  7 ON
VNS  NO
SCNV NO
CNV  NO
EXP  NO
FRL  0
DMI  10
ISDM 0
FCI  0
FSNI 0
BNE  NO
DORG NO
SBOC NRR
IDBB DBD
IOHQ NO
OHQ  NO
CBQ  NO
```

```
ISET 0
NALT 5
MFRL 0
OVLL 0


MEM AVAIL: (U/P): 37549072    USED U P: 523769 95732    TOT: 38168573
DISK SPACE NEEDED: 97 KBYTES
REQ ****
```

c) Create NPA 1613 to route the call to CS1000_A.

```
>ld 90
ESN000

MEM AVAIL: (U/P): 37549072    USED U P: 523769 95732    TOT: 38168573
DISK SPACE NEEDED: 97 KBYTES
REQ  prt
CUST 0
FEAT net
TRAN ac2
TYPE npa

NPA  1613

NPA  1613
RLI  10
SDRR NONE
ITEI NONE


MEM AVAIL: (U/P): 37549072    USED U P: 523769 95732    TOT: 38168573
DISK SPACE NEEDED: 97 KBYTES
REQ
```

4.10.4. Configuration Details on SPS

4.10.4.1 Create the gateway endpoints on SPS for CS1000_A and CS1000_B

a) Log in to UCM and click on **NRSM** to launch NRS manager

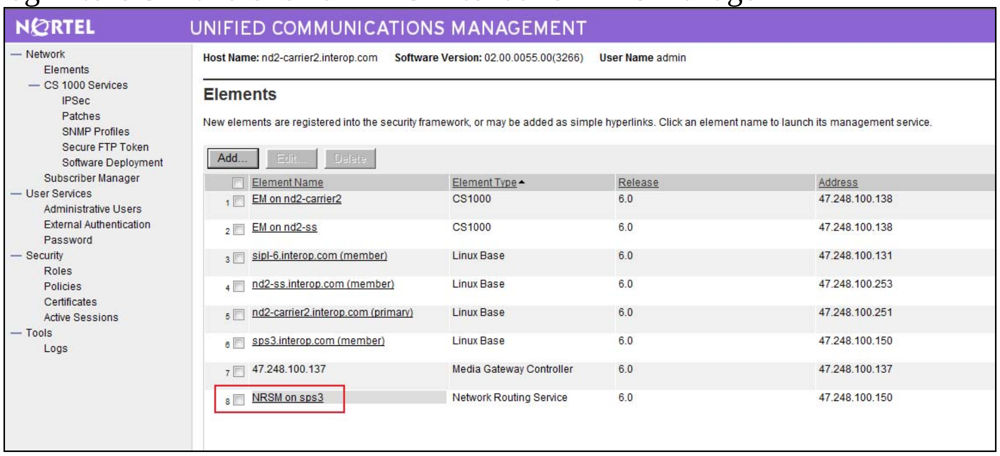

Figure 93 – Launch NRS Manager.

b) Make sure that **Server Domains**, **L1 Domains**, **L0 Domains** have been created.

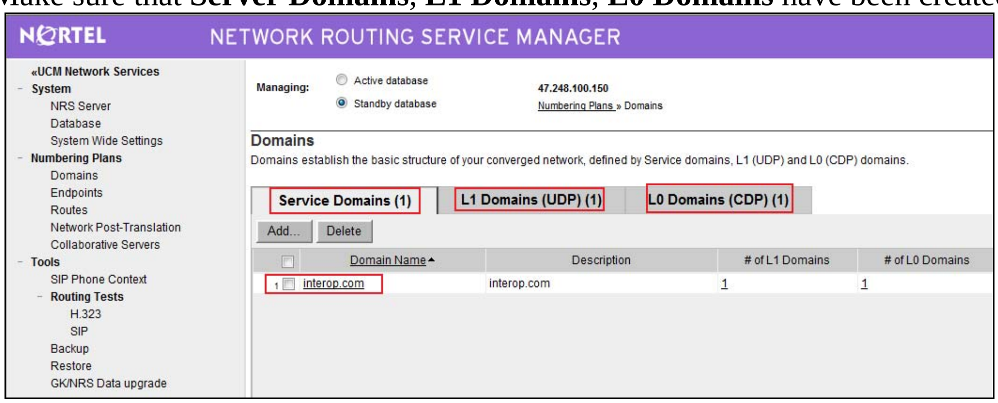

Figure 94 – NRS Manager.

b) Select **Standby database**. Click on **Endpoints**. At the "**Limit results to Domain**", select **Server Domains**, **L1 Domains** and **L0 Domains** and then click **Add** button

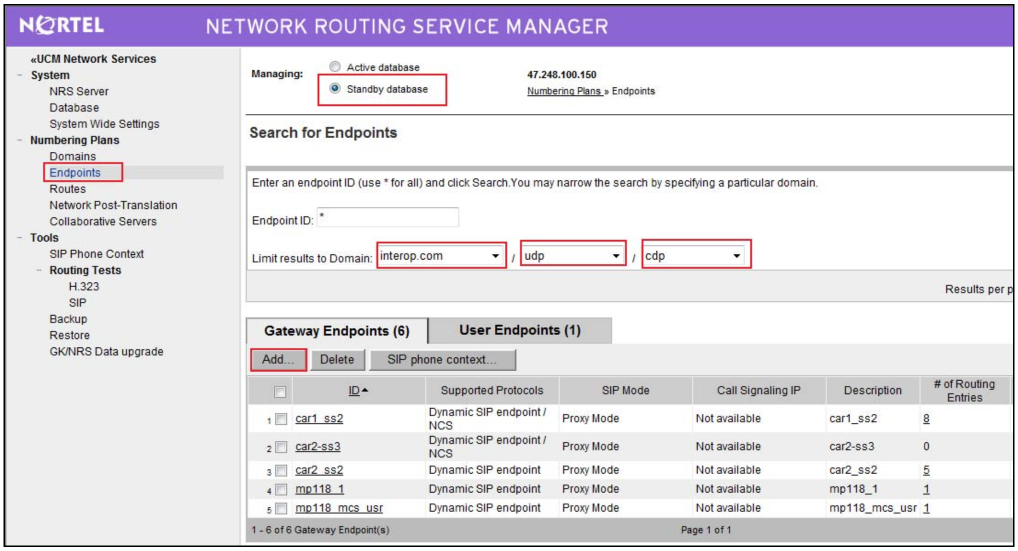

Figure 95 – NRS – Endpoint Creation.

c) Enter **Endpoint name**, **Description**, **country code**, **area code**

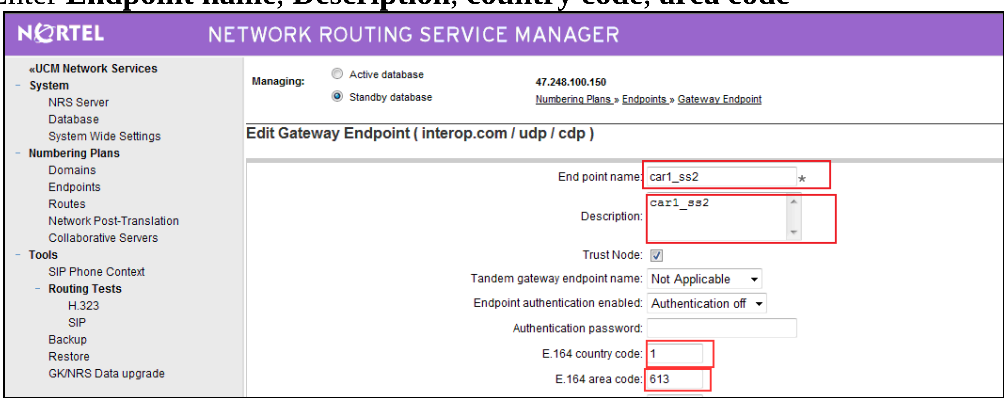

Figure 96 –Gateway Endpoint Configuration Details.

d) For **SIP support**, select **Dynamic SIP endpoint.** Check **SIP UDP transport enabled.** Port is 5060. Check **Network Connection Server enabled**

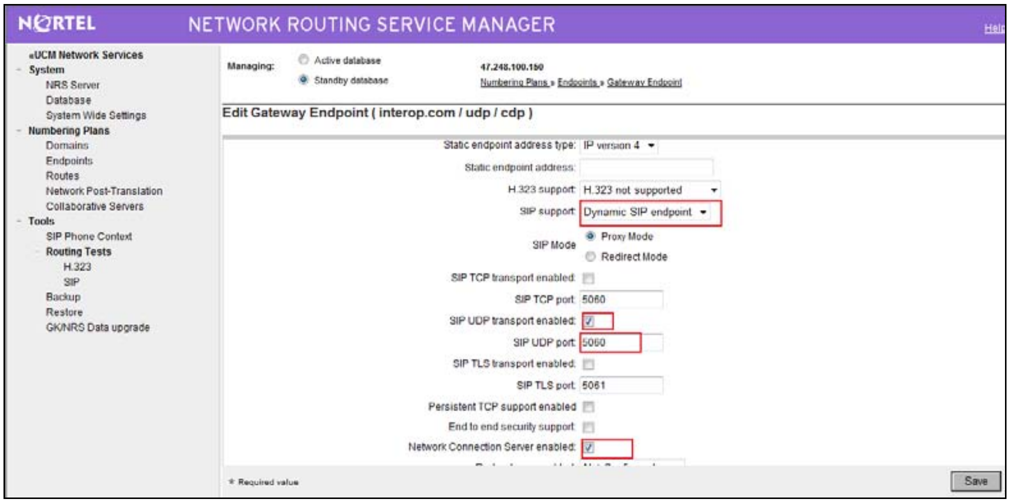

Figure 97 – Gateway Endpoint Configuration Details.

e) Click **Save** to finish.

f) Please do the same to create Endpoint car2_ss2 for CS1000_B. After created 2 endpoints car1_ss2 and car2_ss2, **Figure 98** is showed as follows.

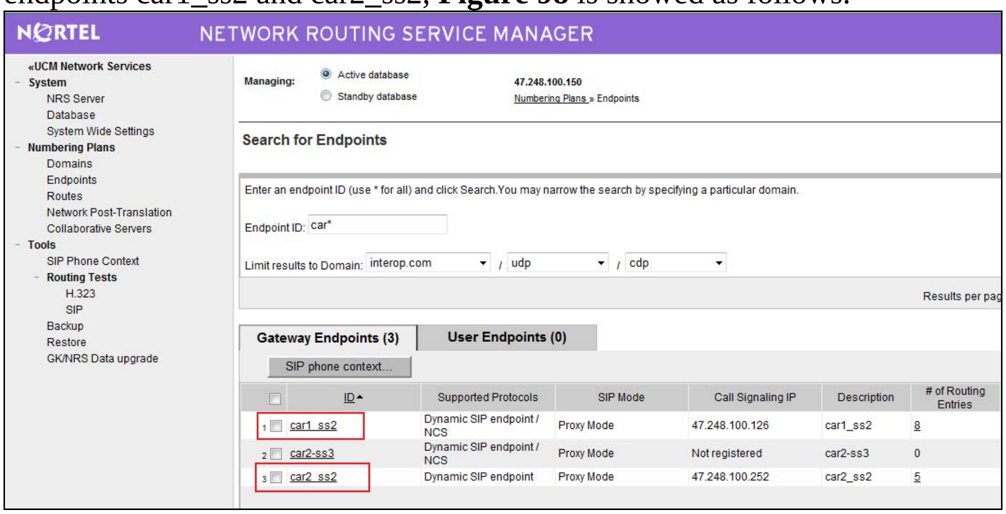

Figure 98 – Gateway Endpoint Details.

4.10.4.2 Create the routing entries for each of gateway endpoints on SPS

a) Select **Standby database**. Click on **Routes**. At "**Limit results to Domain**", select a correct **Server Domains**, **L1 Domains** and **L0 Domains**. At **Endpoint name**, select the endpoint name of CS1000_A (car1_ss2), and then click **Add** button.

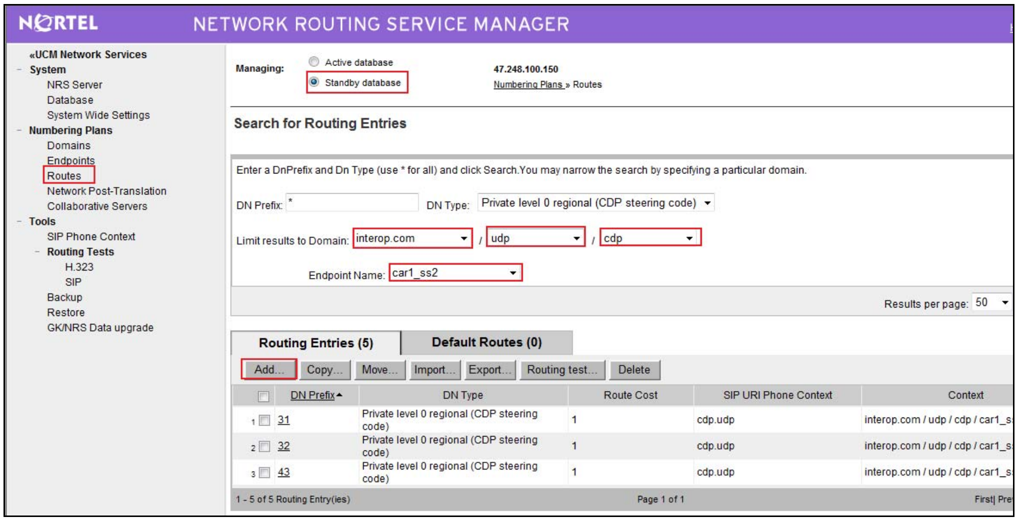

Figure 99 – Routes Configuration Page.

b) Select **DN type** is CDP and **DN prefix** is 53 and then click **Save** button.

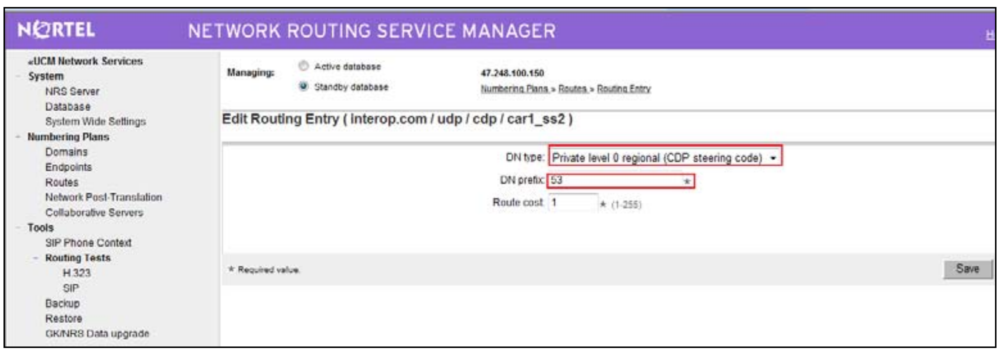

Figure 100 – CS1000_A (car1_ss2) - Routes Configuration Details Page.

c) Do the same to create NPA 1613.

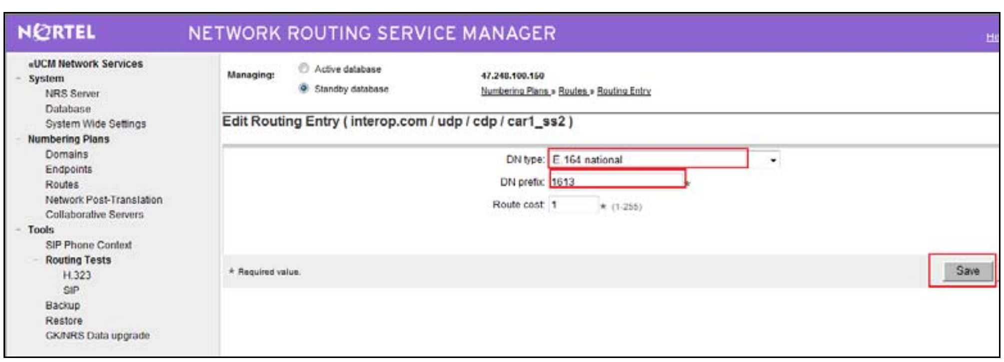

Figure 101 – CS1000_A (car1_ss2) - Routes Configuration Details Page.

After finished, we will have NPA 1613 and CDP 53 for CS1000_A as shown in **Figure 102**.

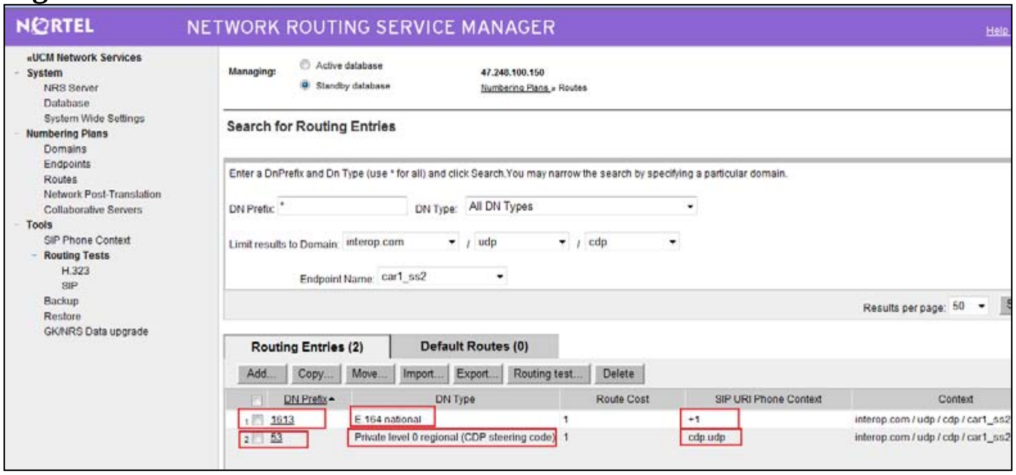

Figure 102 – Routes Entries for CS1000_A (car1_ss2).

d) Do the same to create CDP 50 for CS1000_B. we will have the Routes entries for CS1000_B as shown in **Figure 103**.

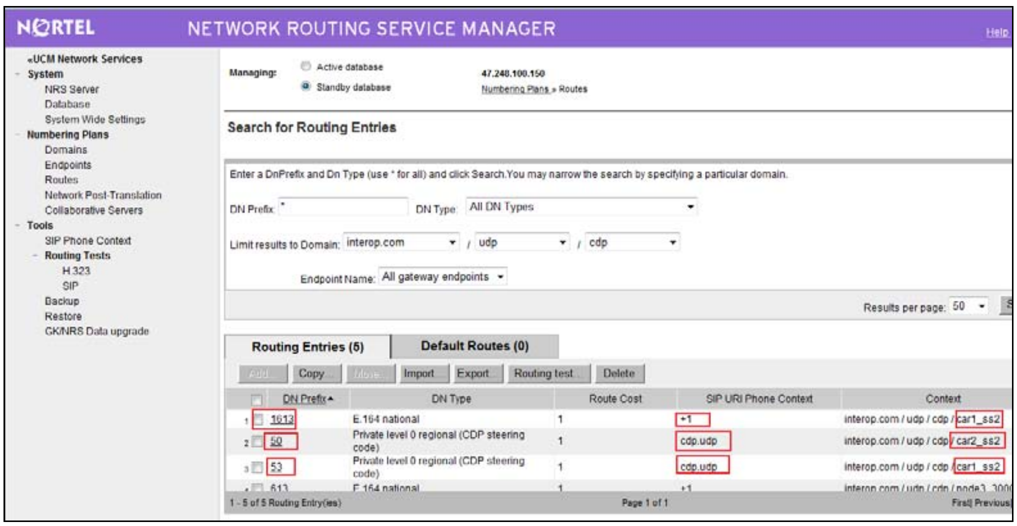

Figure 103 – Routes Entries for CS1000_A (car1_ss2).

4.10.4.3 Save Configuration

a) Click on **Database**, and then click **Cut Over** as shown in **Figure 104**.

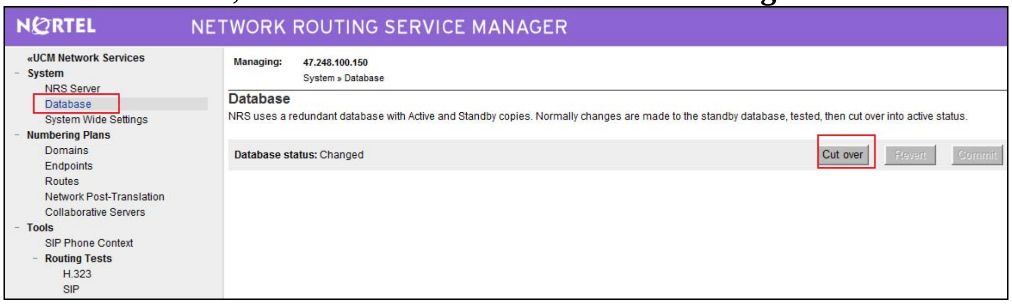

Figure 104 – Database Cut Over.

b) Click on **Commit** as shown in **Figure 105**.

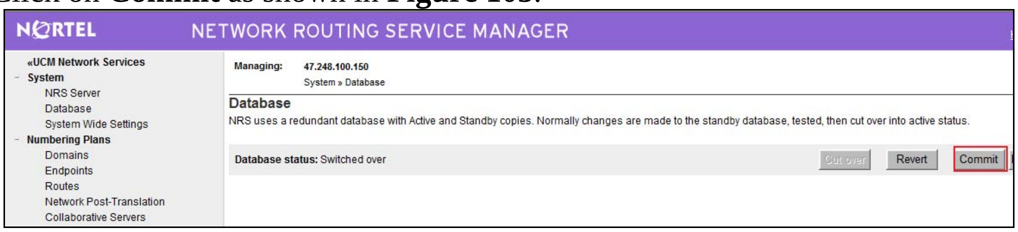

Figure 105 – Database Commit.

b) Click on **Endpoints**, select **Active database** as shown in **Figure 106**.

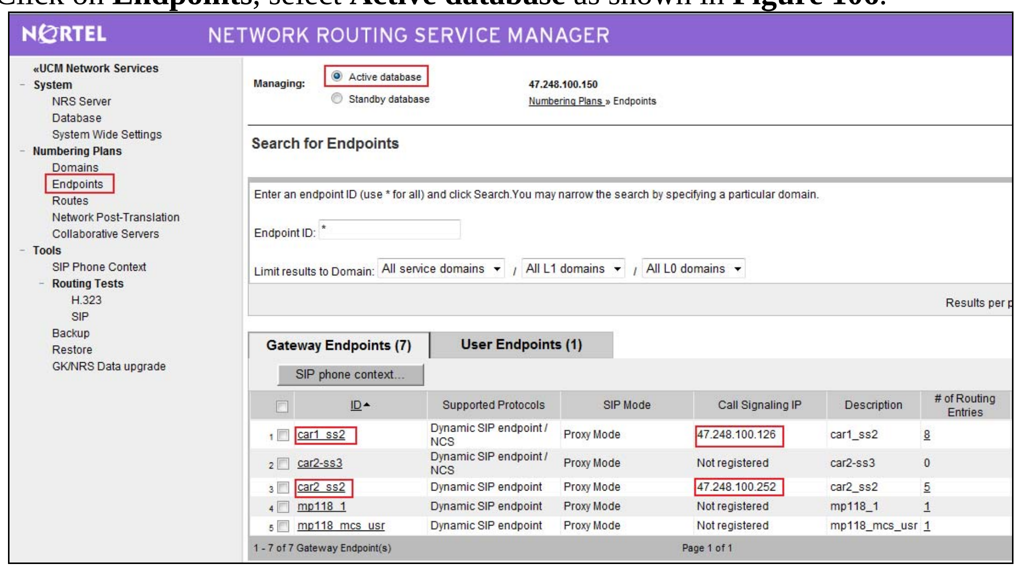

Figure 106 – Active Database.

5. Skype Connect Service configuration

Here is a summary,

5.1 Registering for Skype Manager. https://login.skype.com/bcp/login?message=login_required
5.2 Creating a SIP Profile.
5.3 Buying Channel Subscriptions
5.4 Allocating Skype Credit for outbound calling.
5.5 Inbound Calls:
 5.5.1 Setting up a SIP Profile for inbound calling using business accounts.
 5.5.2 Setting up a SIP Profile for inbound calling using Online Numbers.
5.6 Outbound Calls: Setting up Caller IDs for a SIP Profile.
 5.6.1 Setting up a Caller ID using a landline number
 5.6.2 Setting up a Caller ID using an Online Number

Important: we can not call people on Skype from CS1000.

Skype user guide can be found at the link "**http://download.skype.com/share/business/guides/skype-for-sip-user-guide.pdf**"

5.1. Registering for Skype Manager

Before you can get started with Skype for SIP, you need to register to use Skype Manager. To do so, visit Skype for Business website skype.com/business and click Skype Manager. Follow the instructions to specify the personal Skype account you want to use to set up Skype Manager.

Step 1: Select **Business** -> **Sign into Skype Manager**, and click **Register**.

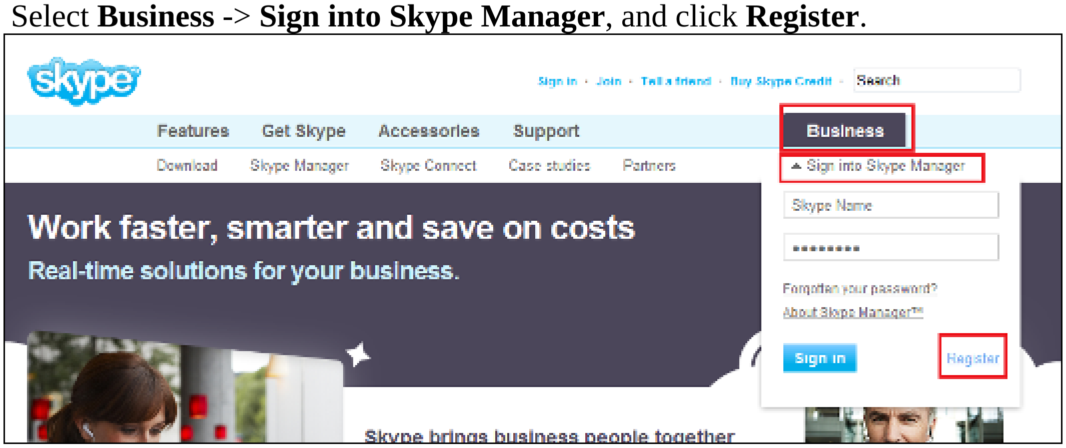

Figure 107 – Skype Registration

Step 2: Click on "**Sign up now**"

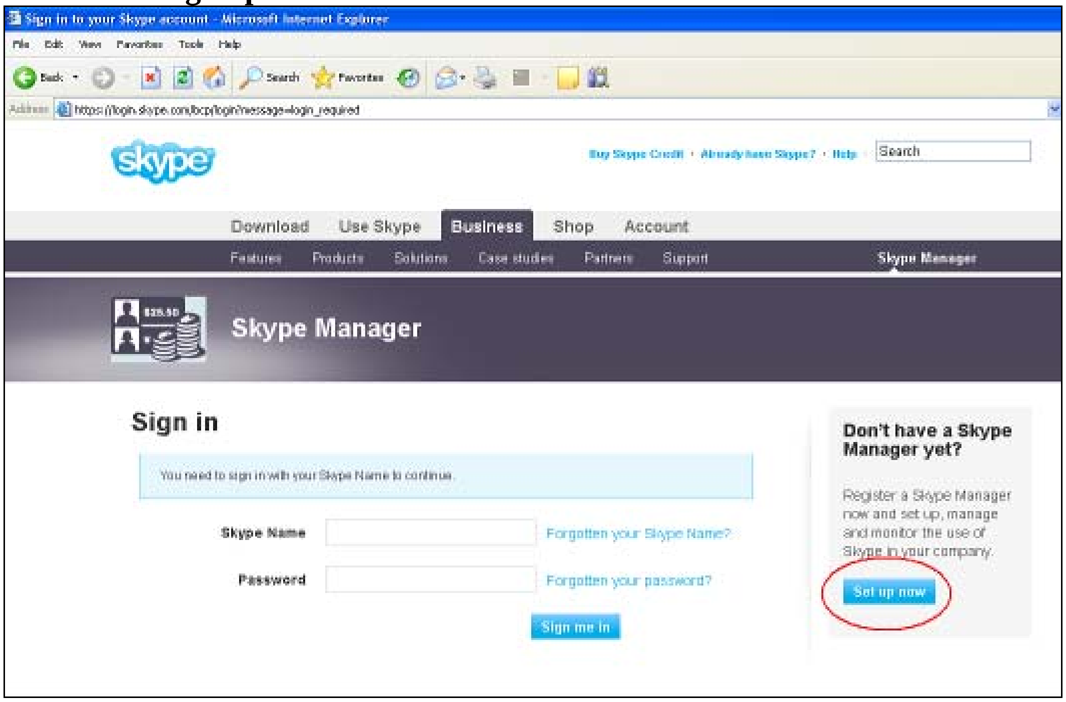

Figure 108 – Skype Registration (cont.)

Step 3: Select "**No, I don't have a Skype account**".

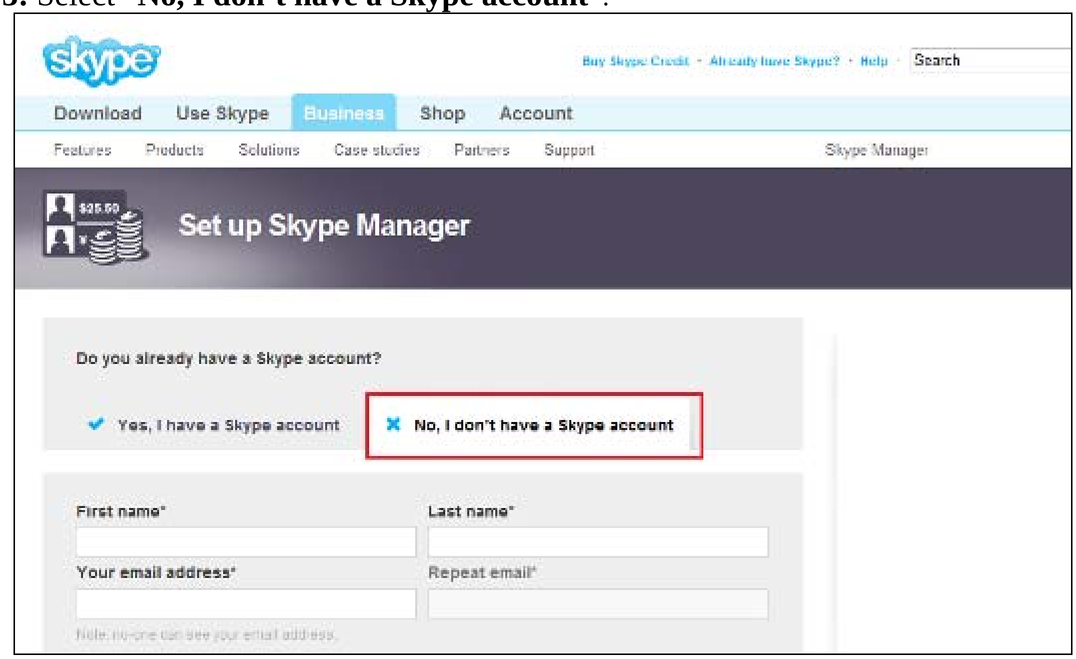

Figure 109 – Skype Registration (cont.)

Step 4: After set up Skype manager completes, log in to Skype Manager, and then click on **Feature** to create **SIP profile**

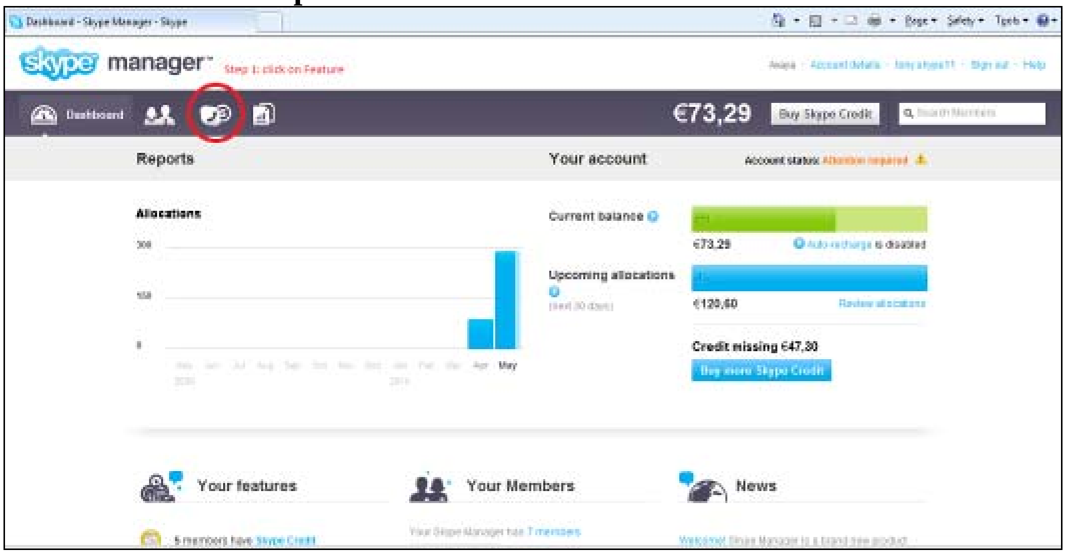

Figure 110 – Skype Registration (cont.)

5.2. Creating a SIP Profile

To use Skype Connect, you need to create at least one SIP Profile in Skype Manager. Creating a SIP Profile is straightforward, using the process described in Section 5.0 - Creating a SIP Profile, page 10 in Skype user guide**.**

Note: regarding to new Skype manager account, user needs to download Skype software-> Log in to Skype client so Skype Manager will be allowed to create Skype profile.

A SIP Profile comprises of six elements,

1. SIP Credentials: these are the log in details needed by CS1000 to connect to Skype.
2. Skype Credit: to pay for outbound calls, if required.
3. Business accounts: for receiving calls from Skype in CS1000, if required.
4. Online Numbers: so people can call from landlines and mobiles, which will be directed to CS1000, if required.
5. A monthly Channel Subscription: which determines the number of concurrent calling channels you want to use with Skype for SIP.
6. Your preferred Caller ID: which can be any Online Number you have associated with your SIP Profile or a landline number your company is authorized to use once it has completed Skype's company verification process.

A SIP Profile may be configured for outgoing calls:

To enable a SIP Profile for outgoing calls, you must allocate Skype Credit to this SIP Profile from your Skype Manager. If the Caller ID option has been set up for this SIP Profile, then it may be used for the outgoing calls going out from a phone without Caller ID configured.

A SIP Profile may also be configured for incoming calls,

To enable incoming calls, choose one or more business accounts to be associated with the SIP Profile. You can also purchase Online Numbers and associate them to your SIP Profile. Incoming calls to those business accounts or Online Numbers will be directed to your SIP Profile.

Do not use the same SIP Profile on more than one CS1000.

You may create as many SIP Profiles as you want in Skype Manager. This is useful if you,

- Have multiple CS1000 SIP Gateway. It means you can create a separate SIP Profile for each SIP Gateway.
- Wish to separate and manage outbound calling costs from different parts of your organization.

After created, we have a SIP Profile 1 as follows.

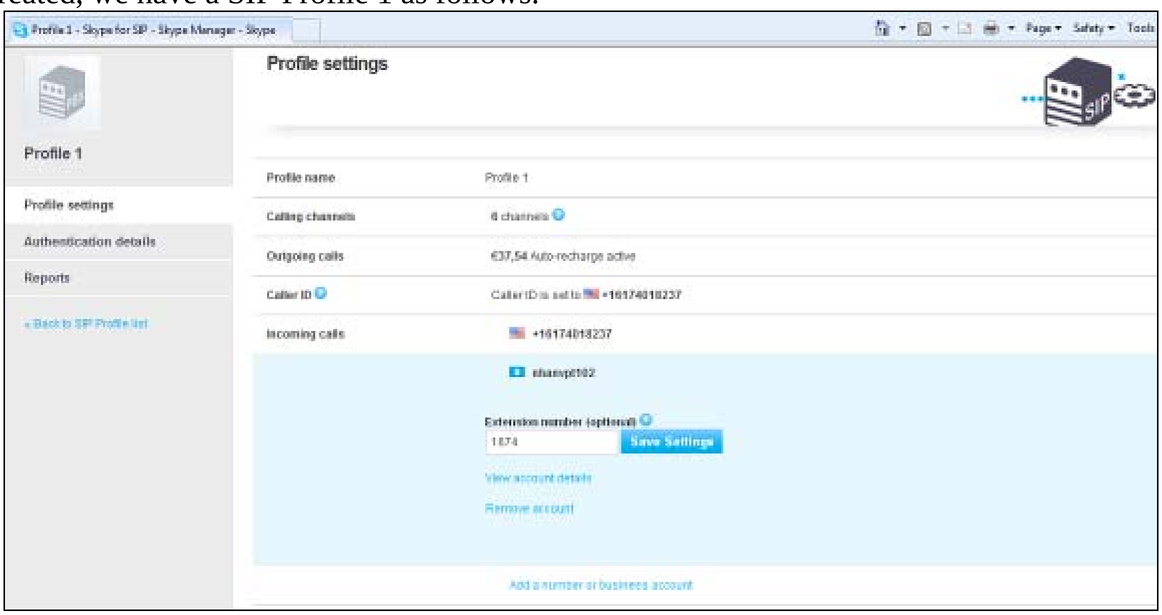

Figure 111 – Skype Profile Settings

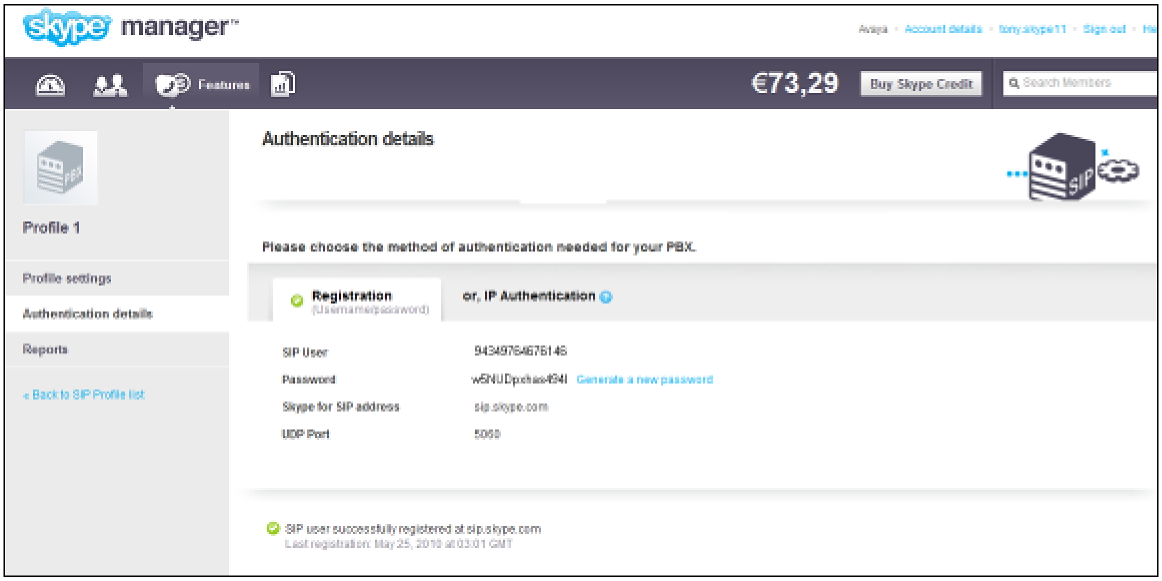

Figure 112 – Authentication Details

5.3. Buying Channel Subscriptions

Channel subscriptions are the amount of concurrent calling channels you would like to use with your SIP Profile and are charged on a monthly basis. Skype for SIP supports up to 300 simultaneous calling channels, enabling up to 300 concurrent conversations. Please follow Section 6.1 Buying Channel Subscriptions, Page 13 in Skype user guide.

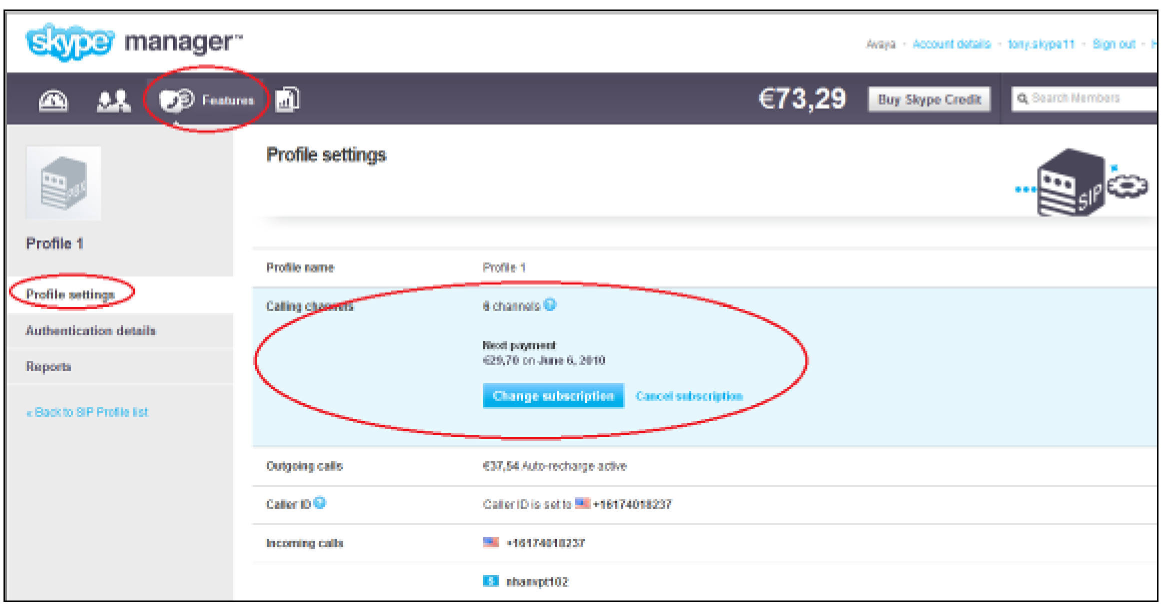

Figure 113 – Subscription in Profile Settings

5.4. Allocating Skype Credit for Outbound calling

In Skype Manager, Skype calls are normally paid for by Skype Credit being allocated to business accounts. However, Skype for SIP is different because Skype Credit can be allocated directly to a SIP Profile. Skype Credit allocated to a SIP Profile is used only to pay for outbound calls. Channel Subscriptions and fees for Online Numbers are paid directly from your Skype Manager.

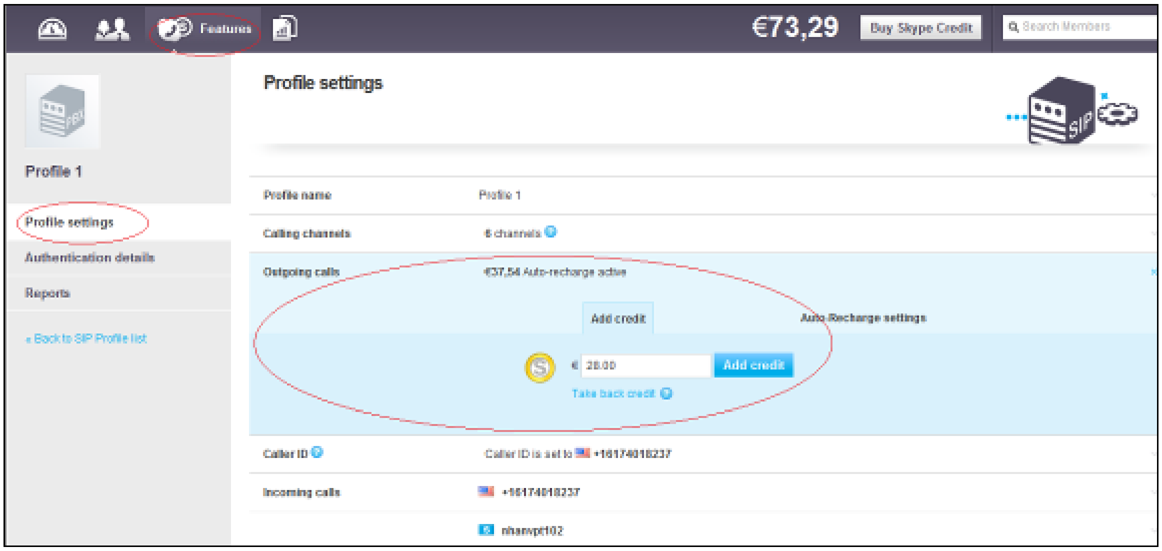

Figure 114 – Account Credit for Outbound Calling

5.5. Inbound Calls

To set up a SIP Profile for inbound calling, you must either,

- Associate one or more business accounts with the SIP Profile.
- Or, assign one or more Online Numbers to the SIP Profile.

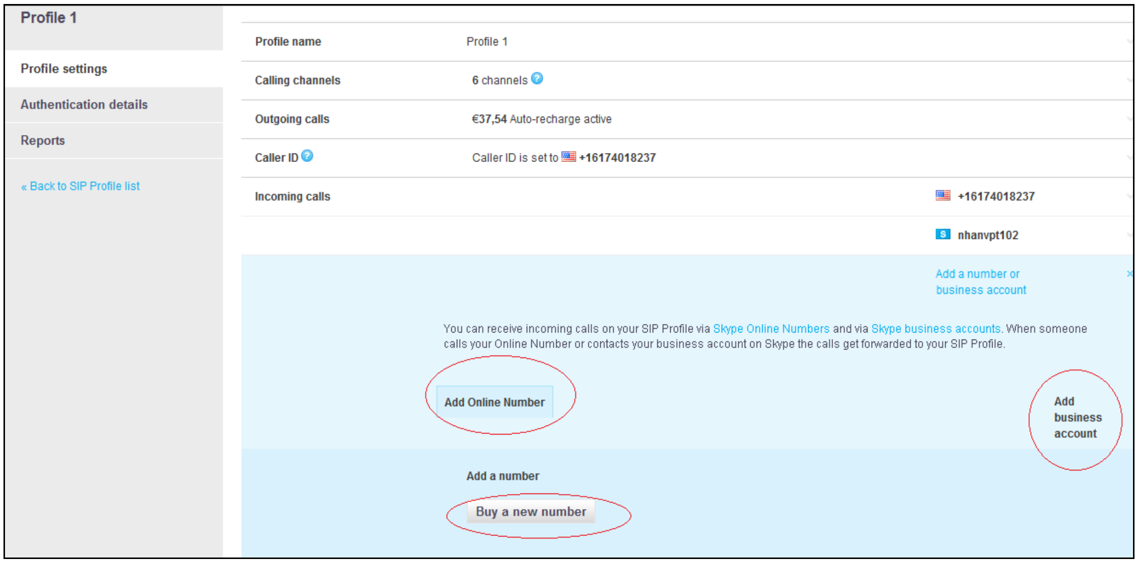

Figure 115 – SIP Profile for Inbound Calling

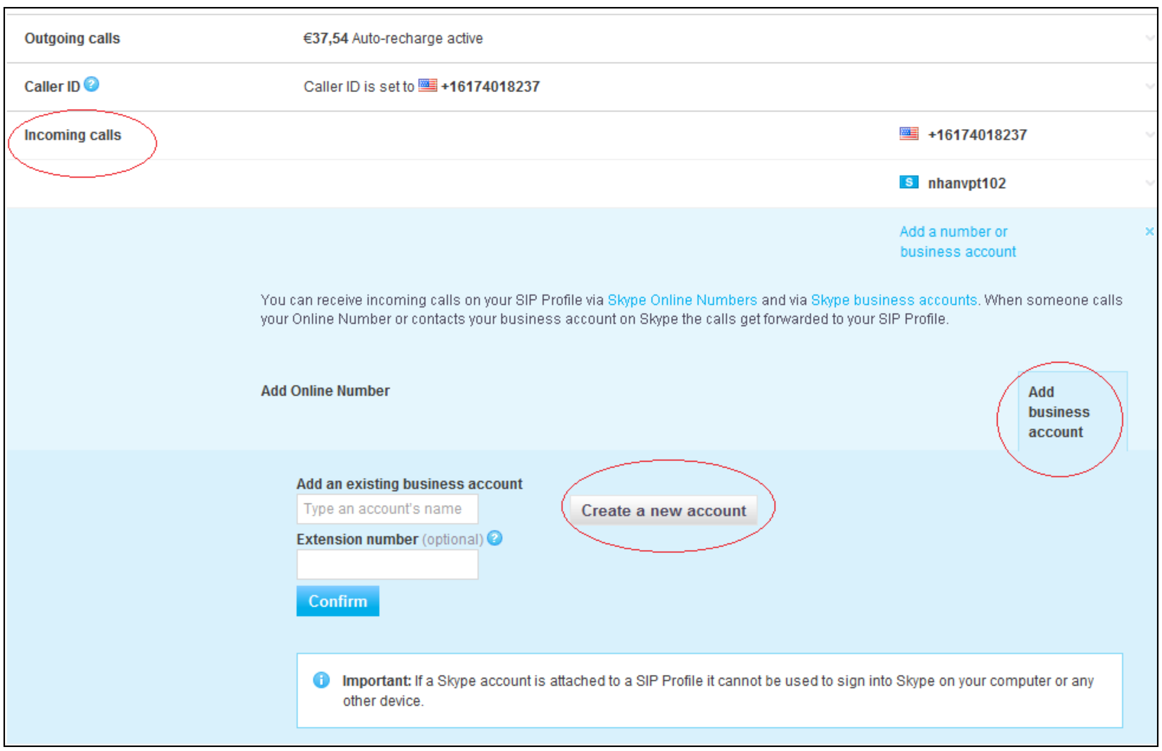

Figure 116 – SIP Profile for Inbound Calling (cont.)

After completed, we have one business account and one online number as follows.

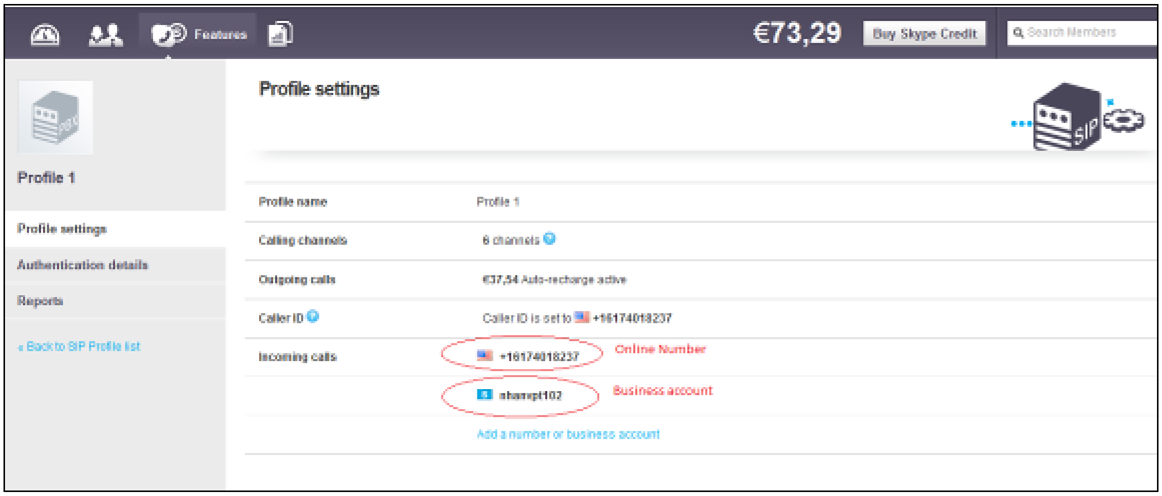

Figure 117 – SIP Profile for Inbound Calling (cont.)

5.6. Outbound Calls

For any SIP Profile, you have three Caller ID options,

- Set Caller ID to be a landline number used by your company (provided your company has been verified by Skype).
- Set Caller ID to be any Online Number that you have assigned to the SIP Profile.
- Choose not to present a Caller ID.

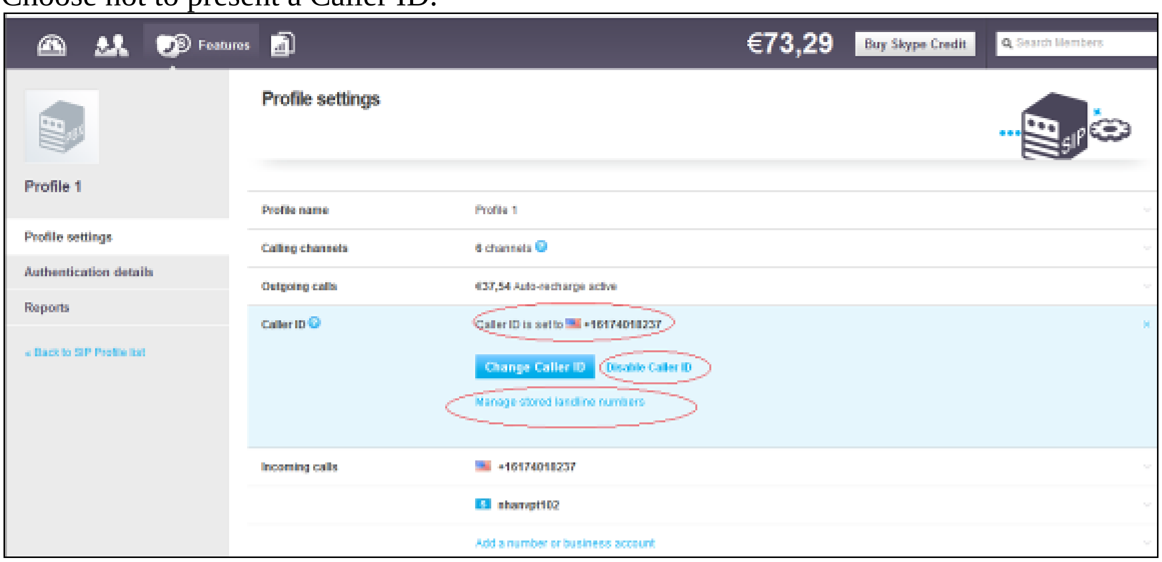

Figure 118 – SIP Profile for Outbound Calling

6. General Test Approach and Test Results

The focus of this interoperability testing was to verify the SIP trunk connectivity between Skype Connect and Avaya Communication Server 1000 release 6.0. The testing verified SIP signaling and media of the basic telephony features are communicating correctly. The following features were covered: basic calls, busy, music on hold, blind and consultative transfers, DTMF, MWI, codec negotiations, conference, voice mail, caller ID presented to PSTN.

6.1. General test approach

The main objectives were to verify the following,

- Installation, registration and integration with Skype Connect.
- The basic telephony features:
  - Call establishment among Skype clients, Avaya IP Phones (listed in Section 3), digitals, analogs and PSTN phones.
  - Basic call operations: on-hold/ retrieve, blind/consultative transfer, call forward, conference.
  - Caller ID of each phone on Avaya CS1000 can be presented to PSTN.
  - Caller ID restriction.
  - DTMF transmission.
  - Voicemail with MWI notification.
  - Codec negotiation.
- Performance tests:
  - Only tested with 3 calls at the same time.
  - Conference with 6 phones.
  - Call on-hold up to an hour.
  - Call duration up to 4 hours.
- Negative testing:
  - Disable IP connectivity to simulate the drop of SIP trunk.
  - Authentication challenge.
  - Account out of balance.

6.2. Test Results

The objectives outlined in Section 6.1 were verified and met. The following observations were made during interoperability testing,

<u>Issue 01.</u> Dial to telephone number which begins with “*”, i.e. *xxxxx does not match required format on Avaya CS1000.

Issue 02. CPND - Call Party Name Display is not supported on test set up. Telephone number is displayed instead.

Issue 03. Music on hold is not enabled on CS1000. i.e. User won't hear music when call is put on hold

Issue 04. Media Security is not enabled on this test configuration.

Issue 05. Do not change the SIP trunk gateway expires timer on CS1000.

Issue 06. With dynamic registration method, Skype does respond 200 OK to the registration of CS1000 but Skype Manager always displays "SIP user is not yet registered at sip.skype.com" and as such, PSTN can not call to CS1000. There is no INVITE from Skype to CS1000. This issue was fixed on the Amsterdam SBC by Skype.

The dynamic registration was tested against the Amsterdam SBC (IP address is 80.252.85.76). ACME made a configuration change to support the CS1k Registration on the Amsterdam SBC. CS1K sends the contact header as the example below:
Contact:<sip:99051000111476@sip.skype.com:5060;maddr=47.248.100.126;transport=UDP>

ACME modified it to look like:
Contact: <sip:99051000111476@sip.skype.com:5060;transport=UDP>;

Issue 07. Making a phone call from Skype client to a phone on CS1000. The CS1000 phone answers. The call is established and then the CS1000 phone puts Skype client on hold for an hour. User expects that the call stays for the complete hour. However, Skype client drops the call after 4 minutes. This issue was fixed by Skype.

Issue 08. Establish a call between CS1000 phone and PSTN phone#1 and then from CS1000 phone, do blind transfer or consult transfer to PSTN phone#2. The call is transferred successfully. The call between PSTN phone#1 and phone#2 is established but CLID is not updated. PSTN phone#1 and phone#2 always display the number of CS1000 phone. Please refer to CR Q02150266.

Issue 09. Make an abandoned call from CS1000 SIP-Line phone to PSTN phone. PSTN phone rings and then CS1000 SIP-Line phone hangs up before PSTN phone answers. The call is released. However, SIP-Line phone is not able to make any other call right after that. It displays "Temporily unavailable" when making outbound calls. Wait for around 30 second, CS1000 SIP-Line phone will be back to normal status and can make outbound call.

Issue 010. The CS1000 phone can not make any outbound calls when it is assigned to an online number to present to PSTN phones. For more detail, please refer to CR Q02164502. This issue is fixed by the patch MPLR30291 applied on CS1000 SSG.

Issue 011. The CS1000 phone can not make any outbound calls when it is configured with caller ID restriction feature. For more detail, please refer to CR Q02164507. This issue is fixed by the patch MPLR30291 applied on CS1000 SSG.

Issue 012. Skype does not support Diversion header so please make certain that the patch MPLR25529 is not applied. If applied, it will cause a call forwarded to PSTN to fail.

Issue 013. Skype client calls a number on the pbx that is unanswered. Skype sends a 487 request Terminated to CS1000. Skype client sees this as being a busy number and displays two options. Option 1 is to start a redial process and Option 2 is to cancel the call. Both of these options work and neither cause a problem to the CS1000 that is being called or to Skype client. This is a Skype client issue.

Issue 014. Skype client calls an unknown number on the CS1000. The CS1000 sends a 404 when it receives a number that it does not know how to terminate. Skype client sees this as being a busy number and displays two options. Option 1 is to start a redial process and Option 2 is to cancel the call. Both of these options work and neither cause a problem to the CS1000 that is being called or to Skype client. This is a Skype client issue.

Issue 015. Skype client calls to Busy Extension. The CS1000 sends a 486 busy here to Skype. Skype client has a popup that gives a choice to cancel or auto redial but Skype client does not hear busy.

Issue 016. When CS1000 phone places a call on hold to SKYPE network, a re-invite is sent from the CS1000 to SKYPE. The CS1000 does not expect any media from SKYPE network. However, SKYPE network still sends RTPs media to CS1000 phone even though the CS1000 phone placed the call on hold.

7. Verification Steps

This section includes some steps that can be followed to verify the solution is working.

7.1. Verify that calls are established with two-way voice path when making calls from one CS1000 phone to another on the local CS1000.

Verify that IP phones, digital, analog (Fax) register successfully show as below.

Verify status of IP phone registered

```
[nortel@nd1-car1 ~]$ isetShow
=== TPS ===
    1.  Set Information
IP Address      NAT  Model Name   Type  RegType  State      Regd-TN                  FWVsn
47.248.101.117       IP Phone 1120E             1120       Regular online  096-00-01-24  C60
47.248.101.120       IP Phone 2002 Phase 2    2002P2     Regular online  096-00-01-06  DCJ
47.248.101.116       IP Phone 1140E             1140       Regular online  096-00-01-26  C60
47.248.101.115       IP Phone 1220              1220       Regular  online  096-00-01-05 C6O
```

Verify status of digital phone registred.

```
LD 32
Stat 4 0 7
>ld 32
.stat 4 0 7
00 = UNIT 00 = IDLE  (3904)
01 = UNIT 01 = IDLE  (3902)

          ……….
```

Verify status of Analog (Fax machine registered).

```
LD 32
.stat 4 0 8
00 = UNIT 00 = IDLE  (L500)
01 = UNIT 01 = IDLE  (L500)
```

Verify the following basic calls in local CS1000.

IP phone------------------------call--------------------IP phone
IP phone -----------------------call--------------------SIP Line Client
IP Phone -----------------------call--------------------Analog/Fax phone
IP Phone -----------------------call--------------------Digital phone
SIP Line Client----------------call--------------------Analog/Fax phone
SIP Line Client----------------call--------------------Digtal Phone
Analog/Fax phone---------------call--------------------Digital Phone
User can verify the same for calls from oposite direction.

Verify that calls are established with two-way voice path and busy status under CS1000 call server as below.

Verify status of IP phones which are busy

[nortel@nd1-car1 ~]$ isetShow
=== TPS ===

Set Information

IP Address	NAT	Model Name	Type	RegType	State	Regd-TN	UNIStimVsn
47.248.101.117		IP Phone 1120E	1120	Regular	busy	096-00-01-24	C6O
47.248.101.120		IP Phone 2002 Phase 2	2002P2	Regular	busy	096-00-01-06	DCJ
47.248.101.116		IP Phone 1140E	1140	Regular	busy	096-00-01-26	C6O
47.248.101.115		IP Phone 1220	1220	Regular	busy	096-00-01-05	C6O

Verfify status of digital phone is busy

LD 32 .stat 4 0 7 000 = UNIT 00 = BUSY (3904)
01 = UNIT 01 = BUSY (3902)
……

Verify status analog phone is busy

LD 32
.stat 4 0 8
00 = UNIT 00 = BUSY (L500)
01 = UNIT 01 = BUSY (L500)

Verify status of voice gateway if calls are established between IP phone/SIP line Clients to Analog/Digital phones or call to voice message

>>ld 32
NPR000
.stat 4 0 11
00 = UNIT 00 = BUSY (TRK)(IPTN REG)
01 = UNIT 01 = BUSY (TRK)(IPTN REG)
02 = UNIT 02 = BUSY (TRK)(IPTN REG)
03 = UNIT 03 = BUSY (TRK)(IPTN REG)

During the call, use pcap tool (ethereal/wireshark) at the TLAN media gateway card, RTP streams are going for call relate to analog, digital or voice message.

7.2. Verify that calls are established with two-way voice path when making calls from PSTN phone to Avaya phones on the CS1000 through Skype Connect Service via configured SIP trunk.

- Verify basic call between PSTN phones and Avaya phones. At the CS1000 SIP Gateway during the call, use pcap tool (ethereal/wireshark) to make sure that all SIP request/response messages
- Verify Codec, SIP trunk status when call is established under CS1000 call server by tracing DID number

LD 80
.trac 0 496856

ACTIVE VTN 096 0 01 06
ORIG VTN 096 0 01 06 KEY 0 SCR MARP CUST 0 DN 496856 TYPE 2002P2
SIGNALLING ENCRYPTION: INSEC
MEDIA ENDPOINT IP: 47.248.101.120 PORT: 5200
TERM VTN 100 0 00 31 VTRK IPTI RMBR 100 32 OUTGOING VOIP GW CALL
FAR-END SIP SIGNALLING IP: 217.110.230.98
FAR-END MEDIA ENDPOINT IP: 217.110.230.97 PORT: 6478
FAR-END VendorID: Not available
MEDIA PROFILE: **CODEC G.711 A-LAW** PAYLOAD 20 ms VAD OFF
RFC2833: RXPT 101 TXPT 101 DIAL DN 916139675258
MAIN_PM ESTD
TALKSLOT ORIG 21 TERM 53
QUEU NONE
CALL ID 511 941

---- ISDN ISL CALL (TERM) ----
CALL REF # = 416
BEARER CAP = VOICE
HLC =

CALL STATE = 10 ACTIVE
CALLING NO = 442033496856 NUM_PLAN:E164 TON:INTERNATIONAL
ESN:UNKNOWN
CALLED NO = 16139675258 NUM_PLAN:E164 TON:INTERNATIONAL
ESN:UNKNOWN

- Verify SIP Trunk is released when DID number is released the call by tracing that DID number under CS1000 call server

LD 80
.trac 0 496856 (DID number)

- **IDLE** VTN 096 0 01 06 MARP

8. Conclusion

All of the executed test cases have passed and met the objectives outlined in **Section 6.1**, with some exceptions outlined in **Section 6.2**. The outstanding issues are being investigated by Skype and Avaya design teams. Some of these issues are considered as exceptions. Skype Connect Service is considered compliant with Communication Server 1000 release 6.0.

9. Additional References

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

[1] *Communication Server 1000 Network Routing Service Fundamentals, Release 6.0, Revision 01.04, Jun 2009, Document Number NN43001-130*

[2] *IP Peer Networking Installation and Commissioning, Nortel Communication Server 1000 Release 6, Document Number NN43001-313, Version 3.02, May, 2009*

[3] *Communication Server 1000 Overview Release 6.0, Revision 03.04, October 2009, Document Number NN43041-110*

[4] *Communication Server 1000 Unified Communications Management Common Services Fundamentals, Revision 03.05, February 2010, Document Number NN43001-116*

[5] *Communication Server 1000 SIP Line Fundamentals, Release 6.0, Revision 01.08, February 10, Document Number NN43001-508*

[6] *Communication Server 1000 Dialing Plans Reference, Release 6.0, Revision 03.09, June 2009, Document Number NN43001-283*

[7] *Product Compatibility Matrix release 5.0/5.5/6.0, Revision 01.07, February 2010, Document Number NN43001-140*

10. Appendixes

Appendix A: CS1000 CPPM Call Server Rls 6.00R Patches Installed

>ld 143

CCBR000

.mdp issp

VERSION 4021

RELEASE 6

ISSUE 00 R +

DepList 1: core Issue: 03 (created: 2010-06-08 10:00:01 (est))

IN-SERVICE PEPS

PAT#	CR #	PATCH REF #	NAME	DATE	FILENAME	SPECINS
000	Q01981776-01	ISS1:1OF1	p29065_1	27/08/2010	p29065_1.cpm	NO
001	Q00349046-03	ISS1:1OF1	p17588_1	27/08/2010	p17588_1.cpm	NO
002	Q01680019	ISS1:1OF1	p24307_1	27/08/2010	p24307_1.cpm	NO
003	Q01725096-03	ISS1:1OF1	p23200_1	27/08/2010	p23200_1.cpm	NO
004	Q01983521-04	ISS1:1OF1	p27616_1	27/08/2010	p27616_1.cpm	NO
005	Q01849803	ISS1:1OF1	p28064_1	27/08/2010	p28064_1.cpm	YES
006	Q01976701-01	ISS1:1OF1	p28211_1	27/08/2010	p28211_1.cpm	NO
007	Q02024135-04	ISS1:1OF1	p28381_1	27/08/2010	p28381_1.cpm	YES
008	Q02097405	ISS1:1OF1	p24463_1	27/08/2010	p24463_1.cpm	NO
009	Q02029209	ISS1:1OF1	p28469_1	27/08/2010	p28469_1.cpm	NO
010	Q02023636	ISS1:1OF1	p28475_1	27/08/2010	p28475_1.cpm	NO
011	Q02022264	ISS1:1OF1	p28486_1	27/08/2010	p28486_1.cpm	NO
012	Q02030977	ISS1:1OF1	p28507_1	27/08/2010	p28507_1.cpm	NO
013	Q02020526	ISS1:1OF1	p28537_1	27/08/2010	p28537_1.cpm	NO
014	Q02031323-01	ISS1:1of1	p28546_1	27/08/2010	p28546_1.cpm	NO
015	Q02034083	ISS1:1OF1	p28553_1	27/08/2010	p28553_1.cpm	YES
016	Q02028560-04	ISS1:1OF1	p28564_1	27/08/2010	p28564_1.cpm	NO
017	Q02034835	ISS1:1OF1	p28569_1	27/08/2010	p28569_1.cpm	YES

018	Q02033951	ISS1:1OF1	p28579_1	27/08/2010	p28579_1.cpm	NO
019	Q02033139	ISS1:1OF1	p28582_1	27/08/2010	p28582_1.cpm	NO
020	Q01782930-01	ISS1:1OF1	p24964_1	27/08/2010	p24964_1.cpm	NO
021	Q02018384	ISS1:1OF1	p28598_1	27/08/2010	p28598_1.cpm	NO
022	Q02033201	ISS1:1OF1	p28631_1	27/08/2010	p28631_1.cpm	YES
023	Q02089407	ISS1:1OF1	p29311_1	27/08/2010	p29311_1.cpm	NO
024	Q02038675	ISS1:1OF1	p28665_1	27/08/2010	p28665_1.cpm	YES
025	Q02020734-02	ISS1:1OF1	p28668_1	27/08/2010	p28668_1.cpm	NO
026	Q02038440	ISS1:1OF1	p28674_1	27/08/2010	p28674_1.cpm	NO
027	Q02035396	ISS1:1OF1	p28675_1	27/08/2010	p28675_1.cpm	NO
028	Q02038482	ISS1:1OF1	p28682_1	27/08/2010	p28682_1.cpm	NO
029	Q02039994	ISS1:1OF1	p28690_1	27/08/2010	p28690_1.cpm	NO
030	Q02024455-01	ISS1:1OF1	p28717_1	27/08/2010	p28717_1.cpm	NO
031	Q02031359	p28679	p28725_1	27/08/2010	p28725_1.cpm	YES
032	Q02083694	ISS1:1OF1	p29741_1	01/09/2010	p29741_1.cpm	NO
034	Q02108554	ISS1:1OF1	p29534_1	27/08/2010	p29534_1.cpm	NO
036	Q01974383-02	ISS1:1OF1	p27378_1	27/08/2010	p27378_1.cpm	NO
037	Q02092594	ISS1:1OF1	p27830_1	27/08/2010	p27830_1.cpm	NO
038	Q01999478-01	ISS1:1OF1	p27897_1	27/08/2010	p27897_1.cpm	NO
040	Q02007976-03	ISS1:1OF1	p28028_1	27/08/2010	p28028_1.cpm	NO
041	Q02007476	ISS1:1OF1	p28031_1	27/08/2010	p28031_1.cpm	NO
042	Q02011613-01	ISS1:1OF1	p28108_1	27/08/2010	p28108_1.cpm	NO
043	Q02017013-01	ISS1:1OF1	p28313_1	27/08/2010	p28313_1.cpm	NO
044	Q02097631	ISS1:1OF1	p28328_1	27/08/2010	p28328_1.cpm	NO
045	Q01987270-02	ISS1:1OF1	p28416_1	27/08/2010	p28416_1.cpm	NO
046	Q01938235-05	ISS2:1OF1	p28418_2	27/08/2010	p28418_2.cpm	NO
047	Q02032955-02	ISS1:1OF1	p28529_1	27/08/2010	p28529_1.cpm	NO
048	Q02019323-01	ISS1:1OF1	p28551_1	27/08/2010	p28551_1.cpm	NO
049	Q02100914	ISS1:1OF1	p28597_1	27/08/2010	p28597_1.cpm	NO
050	Q02032155	p28538	p28638_1	27/08/2010	p28638_1.cpm	YES
051	Q02040015	ISS1:1OF1	p28657_1	27/08/2010	p28657_1.cpm	NO
052	Q02031118	ISS1:1OF1	p28680_1	27/08/2010	p28680_1.cpm	NO
053	Q02029228-01	ISS1:1OF1	p28681_1	27/08/2010	p28681_1.cpm	NO
054	Q02043231	ISS1:1OF1	p28712_1	27/08/2010	p28712_1.cpm	NO

055	Q02041981	p28695_1	p28719_1	27/08/2010	p28719_1.cpm	NO
056	Q02031959	ISS1:1OF1	p28728_1	27/08/2010	p28728_1.cpm	NO
057	Q02033000	ISS1:1of1	p28736_1	27/08/2010	p28736_1.cpm	NO
058	Q02039217-03	ISS1:1OF1	p28760_1	27/08/2010	p28760_1.cpm	NO
059	Q02043669	ISS1:1OF1	p28771_1	27/08/2010	p28771_1.cpm	NO
060	Q02021470-02	ISS1:1OF1	p28776_1	27/08/2010	p28776_1.cpm	NO
061	Q02033321	ISS1:1OF1	p28801_1	27/08/2010	p28801_1.cpm	NO
062	Q02035555	ISS1:1OF1	p28814_1	27/08/2010	p28814_1.cpm	NO
063	Q02049121-01	ISS1:1OF1	p28819_1	27/08/2010	p28819_1.cpm	NO
064	Q01986974-05	ISS1:1OF1	p28821_1	27/08/2010	p28821_1.cpm	YES
065	Q02031502	ISS1:1OF1	p28832_1	27/08/2010	p28832_1.cpm	YES
066	Q02039427-02	ISS1:1OF1	p28849_1	27/08/2010	p28849_1.cpm	NO
067	Q02095838	1SS1:1OF1	p28852_1	27/08/2010	p28852_1.cpm	NO
068	Q02036885-02	ISS1:1OF1	p28857_1	27/08/2010	p28857_1.cpm	NO
069	Q02043398	ISS1:1OF1	p28869_1	27/08/2010	p28869_1.cpm	NO
070	Q02055997	ISS1:1OF1	p28895_1	27/08/2010	p28895_1.cpm	NO
071	Q02044341	ISS1:1OF1	p28957_1	27/08/2010	p28957_1.cpm	NO
072	Q02058567-01	ISS1:1OF1	p28965_1	27/08/2010	p28965_1.cpm	NO
073	Q02048680	ISS1:1OF1	p28983_1	27/08/2010	p28983_1.cpm	NO
074	Q02062206-01	ISS1:1of1	p28994_1	27/08/2010	p28994_1.cpm	NO
075	Q02063326	ISS1:1OF1	p29027_1	27/08/2010	p29027_1.cpm	NO
076	Q02041385-02	ISS1:1OF1	p29032_1	27/08/2010	p29032_1.cpm	NO
079	Q02043226-02	ISS1:1OF1	p29125_1	27/08/2010	p29125_1.cpm	NO
080	Q02074796	ISS1:1OF1	p29126_1	27/08/2010	p29126_1.cpm	NO
081	Q02084339-02	ISS1:1OF1	p29137_1	27/08/2010	p29137_1.cpm	NO
082	Q02076740	ISS1:1OF1	p29154_1	27/08/2010	p29154_1.cpm	NO
083	Q02071451	ISS1:1OF1	p29164_1	27/08/2010	p29164_1.cpm	NO
084	Q02077171	ISS1:1OF1	p29169_1	27/08/2010	p29169_1.cpm	NO
086	Q02064503	ISS1:1OF1	p29196_1	27/08/2010	p29196_1.cpm	NO
087	Q02073690	ISS1:1OF1	p29208_1	27/08/2010	p29208_1.cpm	NO
088	Q02035822-01	ISS1:1OF1	p29212_1	27/08/2010	p29212_1.cpm	NO
090	Q02065521	ISS1:1OF1	p29218_1	27/08/2010	p29218_1.cpm	NO
091	Q02083027	ISS1:1OF1	p29233_1	27/08/2010	p29233_1.cpm	NO
092	Q02079849	ISS1:1OF1	p29238_1	27/08/2010	p29238_1.cpm	NO

093 Q02086333 ISS1:1OF1 p29262_1 27/08/2010 p29262_1.cpm YES

094 Q02077909 ISS1:1of1 p29272_1 27/08/2010 p29272_1.cpm NO

095 Q02077848-01 ISS1:1OF1 p29320_1 27/08/2010 p29320_1.cpm NO

096 Q02092223 ISS1:1of1 p29343_1 27/08/2010 p29343_1.cpm NO

097 Q02093188 ISS1:1OF1 p29352_1 27/08/2010 p29352_1.cpm NO

098 Q02093256-03 ISS1:1OF1 p29354_1 27/08/2010 p29354_1.cpm NO

099 Q02093325 ISS1:1OF1 p29355_1 27/08/2010 p29355_1.cpm NO

100 Q02012100-06 ISS1:1OF1 p29368_1 27/08/2010 p29368_1.cpm NO

101 Q02094012 ISS1:1OF1 p29370_1 27/08/2010 p29370_1.cpm YES

103 Q02089914 ISS1:1OF1 p29406_1 27/08/2010 p29406_1.cpm NO

104 Q02096318 ISS1:1of1 p29423_1 27/08/2010 p29423_1.cpm NO

105 Q02097948 ISS1:1OF1 p29443_1 27/08/2010 p29443_1.cpm NO

106 Q02100965 ISS1:1 OF 1 p29450_1 27/08/2010 p29450_1.cpm NO

107 Q02102219-01 ISS1:1OF1 p29464_1 27/08/2010 p29464_1.cpm NO

109 Q02103928 ISS1:1OF1 p29486_1 27/08/2010 p29486_1.cpm NO

111 Q02104745-01 ISS1:1OF1 p29495_1 27/08/2010 p29495_1.cpm NO

115 Q02109161 ISS1:1OF1 p29536_1 27/08/2010 p29536_1.cpm NO

117 Q02119261 ISS2:1OF1 p29613_2 27/08/2010 p29613_2.cpm NO

122 Q02096730 p29462 p28557 p29676_1 27/08/2010 p29676_1.cpm NO

124 Q02024749-02 ISS1:1OF1 p29680_1 27/08/2010 p29680_1.cpm NO

125 Q02110973 ISS1:1OF1 p29690_1 27/08/2010 p29690_1.cpm NO

128 Q02096711 ISS1:1OF1 p29714_1 27/08/2010 p29714_1.cpm NO

129 Q02114752 ISS1:1OF1 p29718_1 27/08/2010 p29718_1.cpm NO

130 Q02122052 ISS1:1OF1 p29726_1 27/08/2010 p29726_1.cpm NO

131 Q02124953 ISS1:1OF1 p29744_1 27/08/2010 p29744_1.cpm NO

132 Q02100456-01 ISS1:1 OF 1 p29755_1 27/08/2010 p29755_1.cpm NO

133 Q02125731 ISS1:1OF1 p29802_1 27/08/2010 p29802_1.cpm NO

134 Q02108852 ISS1:1OF1 p29825_1 27/08/2010 p29825_1.cpm NO

135 Q02129264 ISS1:1OF1 p29827_1 27/08/2010 p29827_1.cpm NO

136 Q02128131 ISS1:1OF1 p29830_1 27/08/2010 p29830_1.cpm NO

137 Q02111317 ISS1:1OF1 p29844_1 27/08/2010 p29844_1.cpm NO

138 Q02131547 ISS1:1OF1 p29880_1 27/08/2010 p29880_1.cpm NO

139 Q02135191 ISS1:1OF1 p29935_1 27/08/2010 p29935_1.cpm NO

140 Q02137476 ISS1:1OF1 p29962_1 27/08/2010 p29962_1.cpm NO

141 Q02011541-03 ISS1:1OF1 p29998_1 27/08/2010 p29998_1.cpm NO

144 Q02131549 ISS1:1OF1 p30065_1 27/08/2010 p30065_1.cpm NO

MDP>LAST SUCCESSFUL MDP REFRESH :2010-08-26 14:16:18(Local Time)

MDP>USING DEPLIST ZIP FILE DOWNLOADED :2010-06-15 09:38:10(est)

Appendix B: CS1000 CPPM Signaling Server Rls 6.00.18 Patches Installed

```
[nortel@node1-carrier ~]$ pstat
Product Release: 6.00.18.00
In system patches: 6
PATCH#  NAME      IN_SERVICE   DATE     SPECINS  TYPE   RPM
19      p28774_1   Yes        10/08/10  NO      FRU    nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386
20      p28797_1   Yes        10/08/10  NO      FRU    nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386
21      p29703_1   Yes        10/08/10  NO      FRU    nortel-cs1000-shared-ssSubagent-6.00.18-00.i386
22      p25946_1   Yes        10/08/10  NO      FRU    nortel-cs1000-pi-control-1.00.00.00-00.noarch
24      p27159_1   Yes        10/08/10  NO      FRU    nortel-cs1000-pi-control-1.00.00.00-00.noarch
26      p30291_1   Yes        10/08/10  NO      FRU    nortel-cs1000-pi-control-1.00.00.00-00.noarch

In System service updates: 19
PATCH#  IN_SERVICE   DATE      SPECINS   REMOVABLE   NAME
0       Yes        10/08/10   yes      yes       nortel-cs1000-linuxbase-6.00.18.65-03.i386.001
1       Yes        10/08/10   NO       YES       submgr-2.00.02.00-01.i386.000
2       Yes        10/08/10   NO       YES       nortel-cs1000-gk-6.00.18.63-00.i386.000
3       Yes        10/08/10   NO       YES       nortel-cs1000-sps-6.00.18.63-00.i386.000
4       Yes        10/08/10   NO       YES       nortel-cs1000-shared-general-6.00.18.62-00.i386.000
5       Yes        10/08/10   NO       YES       nortel-cs1000-shared-pbx-6.00.18.62-00.i386.000
6       Yes        10/08/10   NO       YES       nortel-cs1000-emWeb_6-0-06.00.18.63-01.i386.001
7       Yes        10/08/10   NO       YES       nortel-cs1000-pd-6.00.18.62-00.i386.000
8       Yes        10/08/10   NO       YES       nortel-cs1000-dmWeb-6.00.18.62-00.i386.001
9       Yes        10/08/10   NO       YES       nortel-cs1000-csmWeb-6.00.18.62-00.i386.001
10      Yes        10/08/10   NO       YES       nortel-cs1000-auth-6.00.18.62-00.i386.000
11      Yes        10/08/10   NO       YES       nortel-cs1000-ISECSH-6.00.18.62-00.i386.000
12      Yes        10/08/10   NO       YES       nortel-cs1000-dbcom-6.00.18.65-01.i386.001
13      Yes        10/08/10   YES      YES       nortel-cs1000-tps-6.00.18.65-07.i386.000
14      Yes        10/08/10   YES      YES       nortel-cs1000-csv-6.00.18.65-04.i386.000
16      Yes        10/08/10   NO       YES       nortel-cs1000-bcc_6-0-6.00.18.65-02.i386.000
17      Yes        10/08/10   NO       YES       nortel-cs1000-cs1000WebService_6-0-6.00.18.65-02.i386.000
18      Yes        10/08/10   NO       YES       nortel-cs1000-ftrpkg-6.00.18.65-02.i386.000
25      Yes        10/08/10   NO       YES       nortel-cs1000-vtrk-6.00.18.65-TMP297.i386.001
[nortel@node1-carrier ~]$
```

Appendix C: Configure SIP trunk in CS1000 using overlays

Procedure summary

This information is provided as a simple summary of tasks to complete when configuring IP Peer Networking, but it does not replace the full details provided in the IP Peer Networking Guide.

No.	Overlay	Element Management	Action
1	LD 97		Define a virtual super loop
2	LD 17	Select Configuration/D-Channel link	Create a virtual D-channel
3	LD 15	Select Configuration/Customer Explorer link	Define the customer to support ISDN
4	LD 16	Select Configuration/Customer Explorer /Add Route	Create a virtual service route
5	LD 14	Select Configuration/Customer Explorer /Add Trunk	Create virtual trunks

Define a virtual superloop

Use Overlay 97

Prompt	Response	Description
REQ	CHG	
TYPE	SUPL	Configuration data block
SUPL	V100	Virtual superloop number (96 - 112 and multiple of 4 for 11C systems.)//CS 1000 not vloop100

Create a virtual D-channel

Use Overlay 17

Prompt	Response	Description
REQ	CHG	
TYPE	ADAN	Configuration data block
ADAN	NEW DCH 100	Add a primary D-Channel port 100
CTYP	DCIP	D-channel is over IP
DES	VIRTUAL_TRK	Description
USR	ISLD	Integrated services signaling link dedicated
IFC	SL1	Interface type is Meridian 1 – Meridian 1
ISLM	4000	Integrated services signaling link maximum
SIDE	USR	Slave to the controller (USR).

RLS	25	X11 software release of far-end.//not need
RCAP	ND2	Name display format 2//not need

Define a customer with ISDN support

Use Overlay 15

Prompt	Response	Description
REQ	NEW	
TYPE	CDB	Customer data block
CUST	0	Customer number
ANAT	1111	ANI Attendant billing number for making ANI calls
ANLD	111	ANI listed directory number
ISDN	YES	Customer is equipped with ISDN.
VPNI	1	Virtual private network identifier//important
PNI	1	Private network identifier.//important

Define a virtual service route

Use Overlay 16

Prompt	Response	Description
REQ	NEW	
TYPE	RDB	Route data block
CUST	0	Customer number
ROUT	100	Route number
DES	VTRK	Designator field for trunk
TKTP	TIE	TIE trunk only, allowed between SL-1
ICOG	IAO	Incoming and outgoing
VTRK	YES	Virtual trunk route
ZONE	0	Zone for codec selection and bandwidth management
NODE	2000	Node ID of signaling server of this route.

PCID	SIP	Protocol ID for this route
ISDN	YES	ISDN option
MODE	ISLD	Route uses ISDN signaling link
DCH	100	D-channel number for this route
PNI	1	Customer private network identifier.
IFC	SL 1	Interface type : Meridian 1 to Meridian 1
NCNA	YES	Network calling name allowed.
NCRD	YES	Network call redirection.
CHTY	BCH	B-channel type.
CTYP	CDP	Coordinated dialing plan

Define virtual trunks

Use Overlay 14

Prompt	Response	Description
REQ	NEW 32	
TYPE	IPTI	IP trunk
TN	100 0 0 0	Virtual card and channel number
DES	VTRK	Designator field for trunk
CUST	0	Customer number
RTMB	100 1	Route number and member number.
STRI	IMM	Start arrangement incoming
STRO	IMM	Start arrangement outgoing
TGAR	1	Trunk group access restriction.
CHID	1	Channel ID for trunk

Appendix D: Sample inbound and outbound SIP Invite

A sample Inbound SIP Invite

```
Request-Line: INVITE sip:13157914465@47.248.100.244:5060;transport=udp SIP/2.0
```

```
From: <sip:16139675281@sip.skype.com>;tag=a4a109cc-13c4-4c73ca7f-1b7008e0-415501ca
To: <sip:13157914465@sip.skype.com:5060>
Call-ID: CXC-436-68081050-a4a109cc-13c4-4c73ca7f-1b7008e0-4082af6c
CSeq: 1 INVITE
Via: SIP/2.0/UDP 204.9.161.164:5060;branch=z9hG4bK-2852c-4c73ca7f-1b7008e0-63f97f3b
Max-Forwards: 30
User-Agent: SipGW 0.3.19
Allow: INVITE,ACK,CANCEL,OPTIONS,BYE
Contact: <sip:16139675281@204.9.161.164:5060;transport=udp>
Content-Type: application/sdp
Content-Length: 265



Status-Line: SIP/2.0 100 Trying
From: <sip:16139675281@sip.skype.com>;tag=a4a109cc-13c4-4c73ca7f-1b7008e0-415501ca
To: <sip:13157914465@sip.skype.com:5060>
Call-ID: CXC-436-68081050-a4a109cc-13c4-4c73ca7f-1b7008e0-4082af6c
CSeq: 1 INVITE
Via: SIP/2.0/UDP 204.9.161.164:5060;branch=z9hG4bK-2852c-4c73ca7f-1b7008e0-63f97f3b
Supported: 100rel,x-nortel-sipvc,replaces,timer
User-Agent: Nortel CS1000 SIP GW release_6.0 version_ssLinux_6.00.18


Status-Line: SIP/2.0 180 Ringing
From: <sip:16139675281@sip.skype.com>;tag=a4a109cc-13c4-4c73ca7f-1b7008e0-415501ca
To: <sip:13157914465@sip.skype.com:5060>;tag=b0fbf1b8-f464f82f-13c4-40030-a71b2-43de77e0-a71b2


Status-Line: SIP/2.0 200 OK
```

A sample outbound SIP Invite

```
Request-Line: INVITE sip:16139675281@sip.skype.com;user=phone SIP/2.0
From: "Skype 1120"<sip:99051000106920@sip.skype.com>;tag=b0fbe360-f464f82f-13c4-40030-a70fe-
2a120f2c-a70fe
To: <sip:16139675281@sip.skype.com;user=phone>
Call-ID: b0ca0718-f464f82f-13c4-40030-a70fe-5ac2cda6-a70fe
CSeq: 1 INVITE
Via: SIP/2.0/UDP 47.248.100.244:5060;branch=z9hG4bK-a70fe-28c962d8-141e4445
Max-Forwards: 70
Supported: 100rel,x-nortel-sipvc,replaces,timer
User-Agent: Nortel CS1000 SIP GW release_6.0 version_ssLinux_6.00.18
P-Asserted-Identity: "Skype 1120"<sip:13157914465@sip.skype.com;user=phone>
Privacy: none
x-nt-e164-clid: +113157914465@sip.skype.com;user=phone
History-Info: <sip:16139675281@sip.skype.com;user=phone>;index=1
Alert-Info: cid:external@sip.skype.com
x-nt-corr-id: 000002880918131808@0019e1e82491-eccec122
Contact: <sip:13157914465@sip.skype.com:5060;maddr=47.248.100.244;transport=udp;user=phone>
```

Status-Line: SIP/2.0 100 Trying
From: "Skype 1120" <sip:99051000106920@sip.skype.com>;tag=b0fbe360-f464f82f-13c4-40030-a70fe-2a120f2c-a70fe
To: <sip:16139675281@sip.skype.com;user=phone>
Call-ID: b0ca0718-f464f82f-13c4-40030-a70fe-5ac2cda6-a70fe
CSeq: 1 INVITE
Via: SIP/2.0/UDP 47.248.100.244:5060;branch=z9hG4bK-a70fe-28c962d8-141e4445
Contact: <sip:16139675281@sip.skype.com:5060;user=phone;maddr=204.9.161.164;transport=udp>

Status-Line: SIP/2.0 180 Ringing
From: "Skype 1120" <sip:99051000106920@sip.skype.com>;tag=b0fbe360-f464f82f-13c4-40030-a70fe-2a120f2c-a70fe
To: <sip:16139675281@sip.skype.com;user=phone>;tag=a4a109cc-13c4-4c73c9cf-1b6d5b2c-3614ff9b

Status-Line: SIP/2.0 200 OK

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