



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

- CS1000 R6.0 software and implementation of latest patches – on June 15, 2010
- 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.

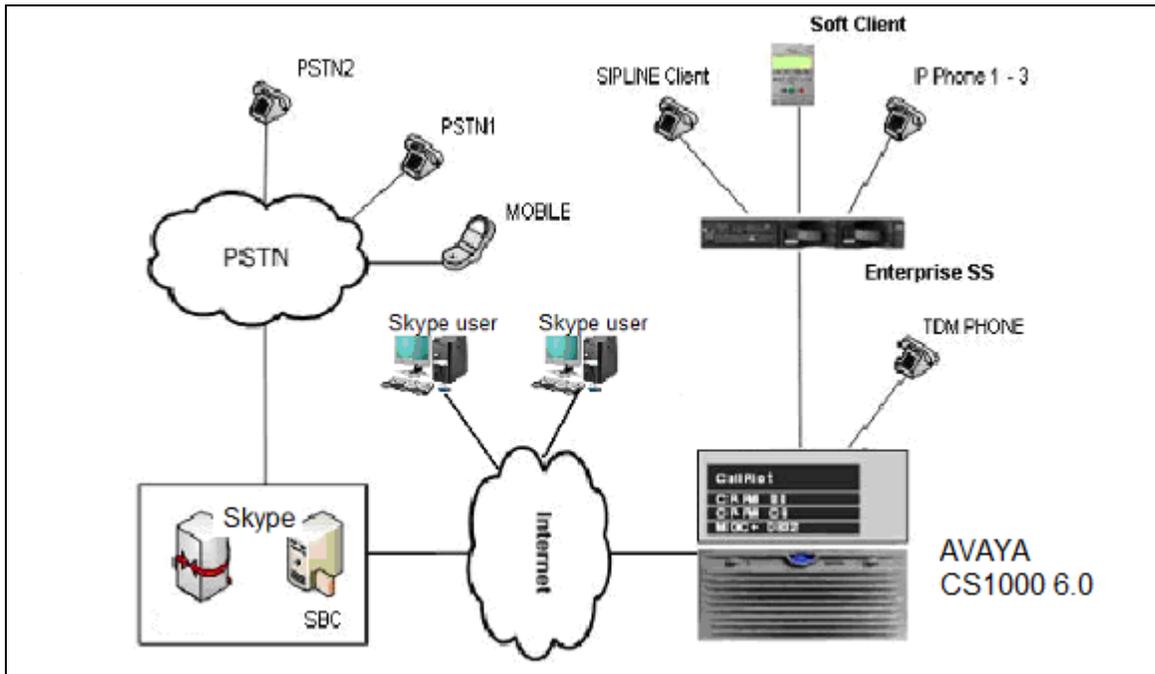


Figure 1- Network diagram for Avaya CS1000 – Skype Connect

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

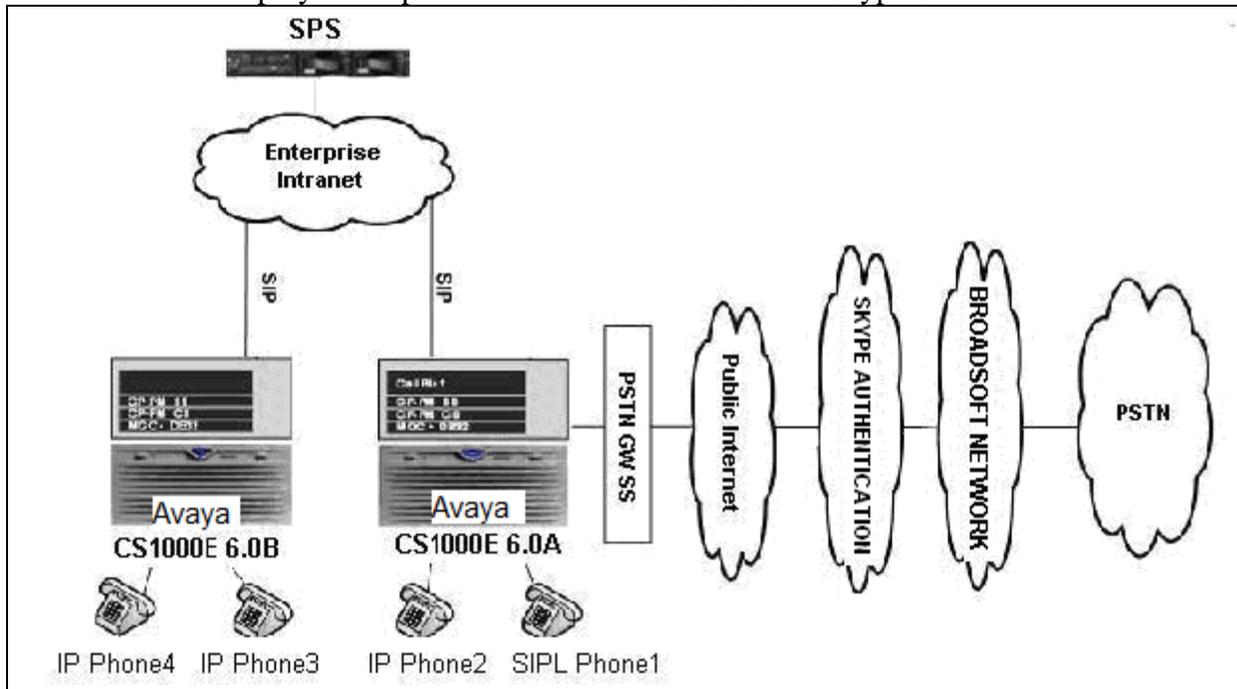


Figure 2 - Network topology for Multi-System configuration for Tandem Calls

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)	<ul style="list-style-type: none"> ● Call Server: 6.00R (CoRes) ● Signaling Server: 6.00.18
Avaya phones	<ul style="list-style-type: none"> ● 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	<ul style="list-style-type: none"> ● 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-6.00.18.00-00.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)
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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)
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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)
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4.6. Administer Dialing Plans

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

- 4.6.6. Inbound Call – Configure IDC to receive the call on an existing phone
- 4.6.7. Outbound National Call
- 4.6.8. Outbound International Call
- 4.6.9. Outbound International Call Restriction

4.7. Phone configuration

- 4.7.1. Calling Line Identification (CLID)
- 4.7.2. IP Phone Creation
- 4.7.3. Outbound Caller ID Restriction

4.8. Administer Voicemail System (Call Pilot) on CS1000

- 4.8.1. Configuration details on CallPilot Manager
- 4.8.2. Configuration detail on CS1000 Call Server

4.9. CS1000 SIP-Line Configuration

- 4.9.1. Configure SIP-Line CS1000 in Element Manager
- 4.9.2. Packages Required for SIP line on CS1000 Call Server
- 4.9.3. Configure SIPL service in LD15
- 4.9.4. Configure DCH for SIPL in LD 17
- 4.9.5. Configure ELAN AML link in LD 17
- 4.9.6. Configure ELAN AML link in LD 17
- 4.9.7. Configure SIPL route
- 4.9.8. Configure SIPL trunks
- 4.9.9. Check status of SIP-Line link and SIP line Gateway
- 4.9.10. Setting password length for SIP line
- 4.9.11. Provisioning SIP client accounts on CS1000 Call Server
- 4.9.12. Check current status of set registration on SLG
- 4.9.13. SMC3456 Softphone Installation
- 4.9.14. Add a SIP Account on SMC3456

4.10. CS1K Tandem Configuration

- 4.10.1. Network topology for multi-system (tandem calls)
- 4.10.2. Avaya Communication Server 1000 A
 - 10.2.1. Configure or add new node IP Telephony
 - 10.2.2. Configure SIP Trunk Gateway
 - 10.2.3. Coordinated Dialing Plan (CDP) - Outbound call to CS1000_B
 - 10.2.4. Coordinated Dialing Plan (CDP) - Inbound call
 - 10.2.5. Configure Dialing Plan - route a call from PSTN to CS1000_B
- 4.10.3. Avaya Communication Server 1000 B
 - 10.3.1. Configure or add new node IP Telephony
 - 10.3.2. Configure SIP Trunk Gateway
 - 10.3.3. Coordinated Dialing Plan (CDP) - Outbound call to CS1000_A
 - 10.3.4. Coordinated Dialing Plan (CDP) - Inbound call

- 10.3.5. Configure Dialing Plan – Outbound call to PSTN via CS1000_A
- 4.10.4. Configuration details on SPS
 - 10.4.1. Create gateway endpoints on SPS
 - 10.4.2. Create the routing entries for each of gateway endpoints on SPS
 - 10.4.3. Save configuration

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.

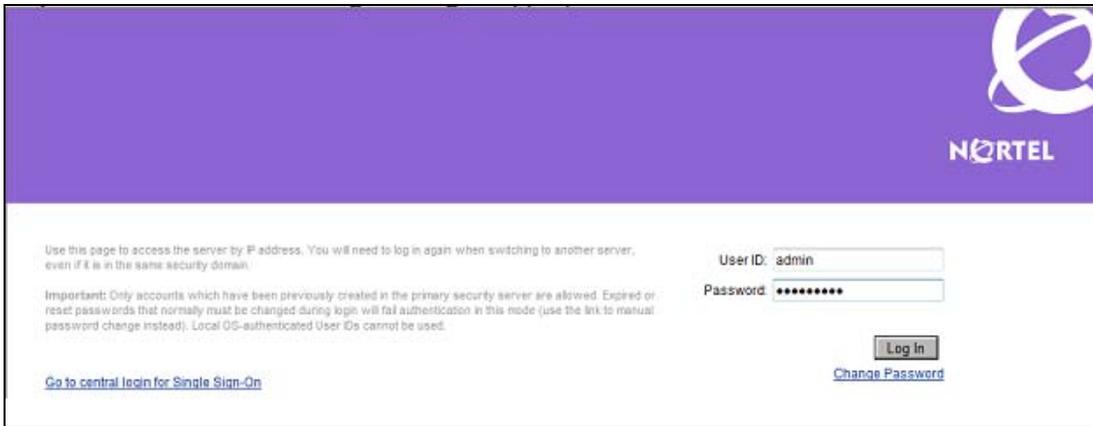


Figure 3 – Log in Unified Communications Management.

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

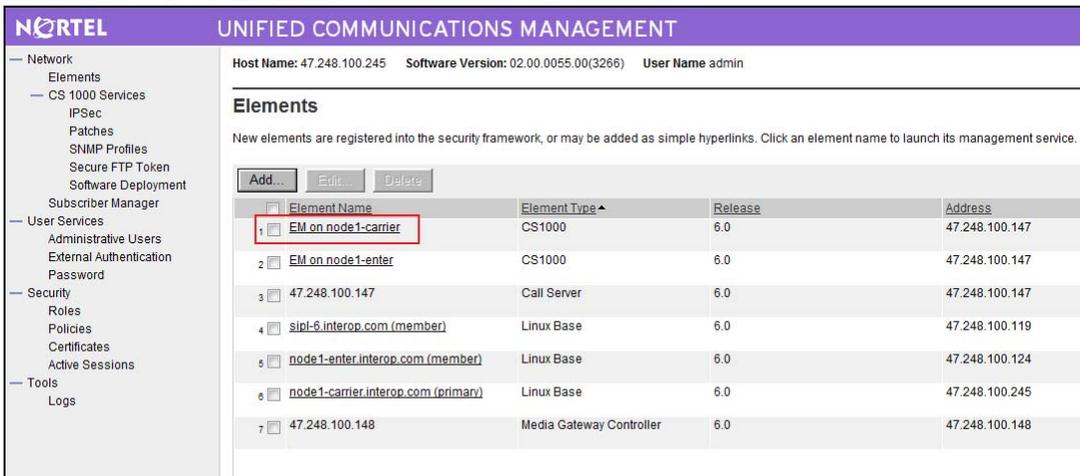


Figure 4 – Unified Communications Management Page.

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

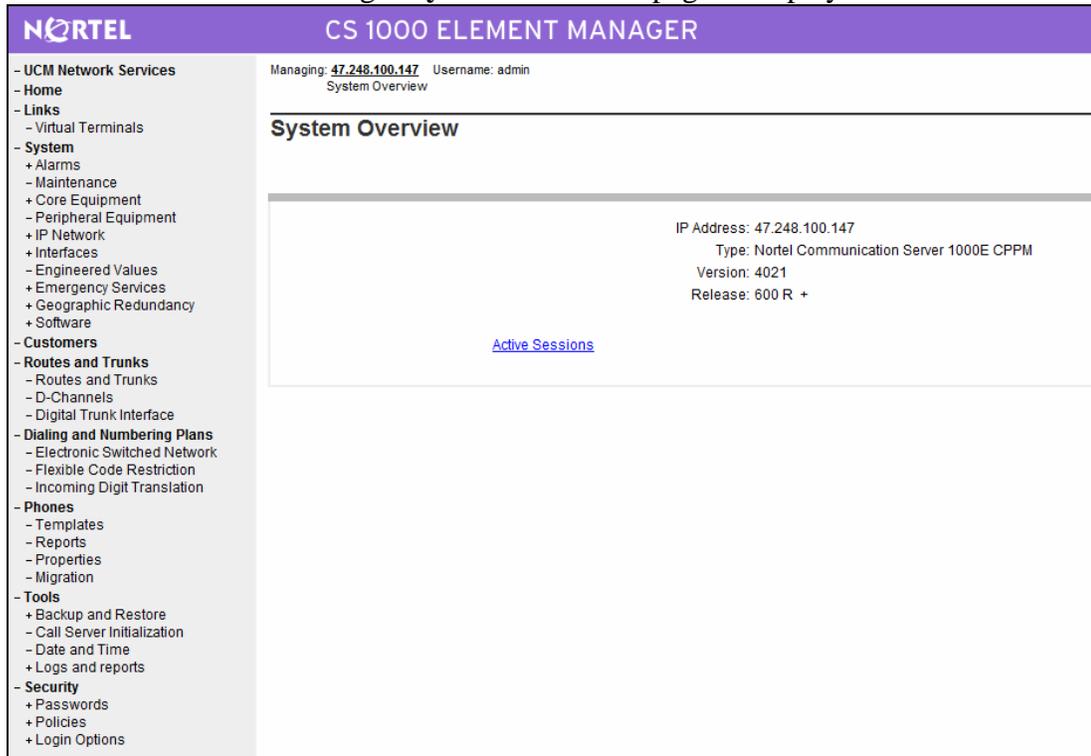


Figure 5 – Element Manager System Overview Page.

Call Server Overlay

- SSH to IP address of SSG or Signaling Server with the nortel/admin account.
- Run the command “cslogin” and log in with the admin account.
- 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.

NORTEL CS 1000 ELEMENT MANAGER

Managing: 47.248.100.147 Username: admin
System » IP Network » IP Telephony Nodes

IP Telephony Nodes
Click the Node ID to view or edit its properties.

Buttons: Add... Import... Export... Delete Print | Refresh

Node ID	Components	Enabled Applications	ELAN IP	TLAN IP	Status
1000	1	PD, Presence Publisher, Gateway (SIPGw)	-	47.248.100.244	Synchronized
1001	1	SIP Line, LTPS, Gateway (SIPGw)	-	47.248.100.126	Synchronized
1002	1	SIP Li	-	47.248.100.120	Synchronized

Show: Nodes Component Servers and Cards

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.

NORTEL CS 1000 ELEMENT MANAGER

Managing: 47.248.100.147 Username: admin
System » IP Network » IP Telephony Nodes

Node Details (ID: 1000 - PD, Presence Publisher, Gateway (SIPGw))

Node ID: 1000 * (0-9999)

Call Server IP Address: 47.248.100.147 *

Telephony LAN (TLAN)

Node IP Address: 47.248.100.244

Subnet Mask: 255.255.255.240 *

Embedded LAN (ELAN)

Gateway IP address: 47.248.100.129 *

Subnet Mask: 255.255.255.224 *

IP Telephony Node Properties

- Voice Gateway (VGW) and Codecs
- Quality of Service (QoS)
- LAN

Applications (click to edit configuration)

- SIP Line
- Terminal Proxy Server (TPS)
- Gateway (SIPGw)

* Required Value. Save Cancel

Associated Signaling Servers & Cards

Select to add Add Remove Make Leader Print | Refresh

Hostname	Type	Deployed Applications	ELAN IP	TLAN IP	Role
node1-carrier	Signaling Server	LTPS, Gateway, PD	47.248.100.149	47.248.100.245	Leader

Note: Only server(s) that are not part of any other IP telephony node and deployed application(s) that match the service(s) selected for this node are available in the servers list.

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)**

Managing: 47.248.100.147 Username: admin
System » IP Network » IP Telephony Nodes

Node Details (ID: 1000 - PD, Presence Publisher, Gateway (SIPGw))

Node ID: 1000 * (0-9999)
Call Server IP Address: 47.248.100.147 *
Telephony LAN (TLAN) Node IP Address: 47.248.100.244 *
Subnet Mask: 255.255.255.240 *
Embedded LAN (ELAN) Gateway IP address: 47.248.100.129 *
Subnet Mask: 255.255.255.224 *

IP Telephony Node Properties

- Voice Gateway (VGW) and Codecs
- Quality of Service (QoS)
- LAN

Applications (click to edit configuration)

- SIP Line
- Terminal Proxy Server (TPS)**
- Gateway (SIPGw)

* Required Value. Save Cancel

Associated Signaling Servers & Cards

Select to add Print | Refresh

Hostname	Type	Deployed Applications	ELAN IP	TLAN IP	Role
<input type="checkbox"/> node1-carrier	Signaling Server	LTPS, Gateway, PD	47.248.100.149	47.248.100.245	Leader

Note: Only server(s) that are not part of any other IP telephony node and deployed application(s) that match the service(s) selected for this node are available in the servers list.

Figure 8 –Node Details Page – TPS.

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

Managing: 47.248.100.147 Username: admin
System » IP Network » IP Telephony Nodes

Node ID: 1000 - UNISstim Line Terminal Proxy Server (LTPS) Configuration Details

UNISstim Line Terminal Proxy Server: Enable proxy service on this node

Firmware

IP Address: 0.0.0.0
Full file path: download/firmwar
Server Account/User ID:
Password:

DTLS

DTLS Policy: Off

Options: Client Authentication
 Periodic Re-keying

* Required Value. Save Cancel

Note: Changes made on this page will NOT be transmitted until the Node is also saved.

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)**.

NORTEL CS 1000 ELEMENT MANAGER

Managing: 47.248.100.147 Username: admin
System » IP Network » IP Telephony Nodes

Node Details (ID: 1000 - LTPS, PD, Presence Publisher, Gateway (SIPGw))

Node ID: 1000 * (0-9999)
Call Server IP Address: 47.248.100.147 *
Telephony LAN (TLAN) Node IP Address: 47.248.100.244 *
Subnet Mask: 255.255.255.240 *
Embedded LAN (ELAN) Gateway IP address: 47.248.100.129 *
Subnet Mask: 255.255.255.224 *

IP Telephony Node Properties

- Voice Gateway (VGW) and Codecs
- **Quality of Service (QoS)**
- LAN

Applications (click to edit configuration)

- SIP Line
- Terminal Proxy Server (TPS)
- Gateway (SIPGw)

* Required Value. Save Cancel

Associated Signaling Servers & Cards

Select to add Print | Refresh

Hostname	Type	Deployed Applications	ELAN IP	TLAN IP	Role
<input type="checkbox"/> node1-carrier	Signaling Server	LTPS, Gateway, PD	47.248.100.149	47.248.100.245	Leader

Note: Only server(s) that are not part of any other IP telephony node and deployed application(s) that match the service(s) selected for this node are available in the servers list.

Figure 10 –Node Details Page – QoS.

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

NORTEL CS 1000 ELEMENT MANAGER

Managing: 47.248.100.147 Username: admin
System » IP Network » IP Telephony Nodes

Node ID: 1000 - Quality of Service (QoS)

Diffserv Codepoint (DSCP)

Enable Nortel Automatic QoS:

Control Packets: 40 (0-63)

Voice Packets: 46 (0-63)

VLAN Tagging: 802.1Q Support

802.1Q Bits Value (802.1P): 6 (0-7)

* Required Value. Save Cancel

Note: Changes made on this page will NOT be transmitted until the Node is also saved.

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**.

The screenshot shows the 'Node Details' screen for Node ID 1000. The interface includes a left-hand navigation menu with categories like 'UCM Network Services', 'System', 'Interfaces', 'Customers', 'Routes and Trunks', and 'Dialing and Numbering Plans'. The main content area displays configuration fields for 'Node ID' (1000), 'Call Server IP Address' (47.248.100.147), 'Telephony LAN (TLAN)' (Node IP: 47.248.100.244, Subnet Mask: 255.255.255.240), and 'Embedded LAN (ELAN)' (Gateway IP: 47.248.100.129, Subnet Mask: 255.255.255.224). Below these are sections for 'IP Telephony Node Properties' (Voice Gateway, QoS, LAN) and 'Applications' (SIP Line, Terminal Proxy Server, Gateway). A 'Save' button is highlighted with a red box. Below the configuration fields is a table for 'Associated Signaling Servers & Cards' with columns for Hostname, Type, Deployed Applications, ELAN IP, TLAN IP, and Role. The table contains one entry: 'node1-carrier' (Signaling Server, LTSPS, Gateway, PD, 47.248.100.149, 47.248.100.245, Leader).

Figure 12 – Synchronize the new Configuration – Save.

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

The screenshot shows the 'Node Saved' screen. It displays a message: 'Node ID: 1000 has been saved on the call server. The new configuration must also be transferred to associated servers and media cards.' Below the message are two buttons: 'Transfer Now...' (highlighted with a red box) and 'Show Nodes'. The 'Transfer Now...' button has a tooltip that says: 'You will be given an option to select individual servers, or transfer to all.'

Figure 13 – Synchronize the new Configuration – Transfer.

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

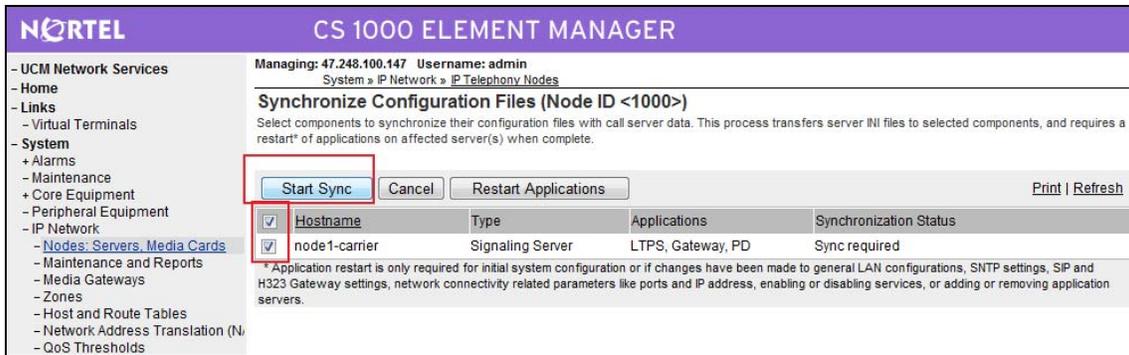


Figure 14 – Synchronize the new Configuration – Start Sync.

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

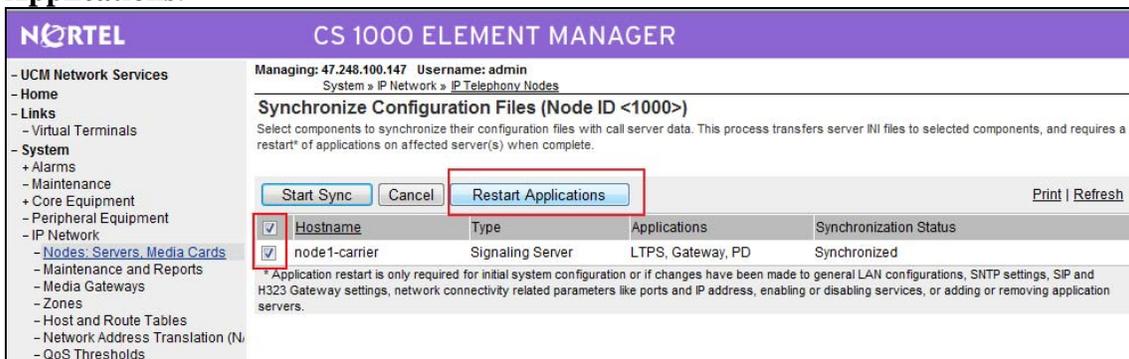


Figure 15 – Synchronize the new Configuration – Restart Applications.

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**.

Managing: 47.248.100.147 Username: admin
System » IP Network » IP Telephony Nodes

Node Details (ID: 1000 - PD, Presence Publisher, Gateway (SIPGw))

Node ID: 1000 * (0-9999)
Call Server IP Address: 47.248.100.147 *
Telephony LAN (TLAN) Node IP Address: 47.248.100.244 * Subnet Mask: 255.255.255.240 *
Embedded LAN (ELAN) Gateway IP address: 47.248.100.129 * Subnet Mask: 255.255.255.224 *

IP Telephony Node Properties

- Voice Gateway (VGW) and Codecs**
- Quality of Service (QoS)
- LAN
- SIP Line
- Terminal Proxy Server (TPS)
- Gateway (SIPGw)

* Required Value. Save Cancel

Associated Signaling Servers & Cards

Select to add Print | Refresh

Hostname	Type	Deployed Applications	ELAN IP	TLAN IP	Role
<input type="checkbox"/> node1-carrier	Signaling Server	LTSPS, Gateway, PD	47.248.100.149	47.248.100.245	Leader

Note: Only server(s) that are not part of any other IP telephony node and deployed application(s) that match the service(s) selected for this node are available in the servers list.

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**.

Managing: 47.248.100.147 Username: admin
System » IP Network » IP Telephony Nodes

Node ID: 1000 - Voice Gateway (VGW) and Codecs

General | **Voice Codecs** | Fax

Codec G711: Enabled (required)
Voice payload size: 20 (milliseconds per frame)
Voice Playback (jitter buffer) delay: 40 80 (milliseconds)
Nominal Maximum
Maximum delay may be automatically adjusted based on Nominal settings.
 Voice activity detection (VAD)

Codec G729: Enabled
Voice payload size: 20 (milliseconds per frame)
Voice Playback (jitter buffer) delay: 40 80 (milliseconds)
Nominal Maximum
Maximum delay may be automatically adjusted based on Nominal settings.
 Voice activity detection (VAD)

Codec G723.1: Enabled
Voice payload size: 30 (milliseconds per frame)

* Required Value. Save Cancel

Note: Changes made on this page will NOT be transmitted until the Node is also saved.

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.

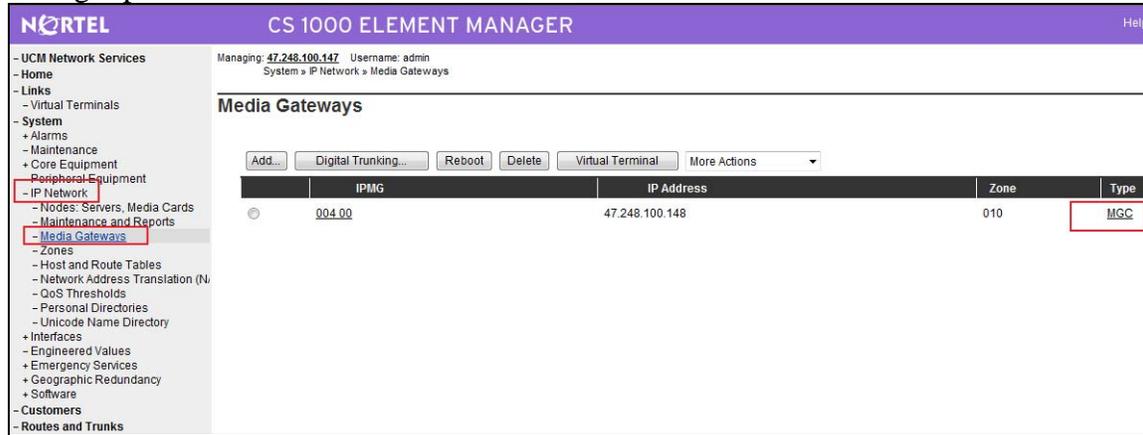


Figure 18 – Media Gateways Configuration Page.

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

NORTEL CS 1000 ELEMENT MANAGER

- VGW and IP phone codec profile

- Enable echo canceller
- Echo canceller tail delay: 64 (milliseconds)
- Enable dynamic attenuation
- Voice activity detection threshold: 1 (0 - 4 DBM)
- Idle noise level: 0 (0 - 1 DBM)
- DTMF tone detection
- Enable low latency mode
- Remove DTMF delay (squelch DTMF from TDM to IP)
- Enable modem/fax pass through mode
- Enable V.21 FAX tone detection
- Fax TCF method: 2
- FAX maximum rate: 14400 (bps)
- FAX payout nominal delay: 100 (0 - 300 milliseconds)
- FAX no activity timeout: 20 (10 - 32000 milliseconds)
- FAX packet size: 30
- + Codec G711 Select
- Codec G729A Select**
- Codec name: G729A
- Voice payload size: 20 (ms/frame)
- Voice playout (jitter buffer) nominal delay: 40
- Modifications may cause changes to dependent settings
- Voice playout (jitter buffer) maximum delay: 80
- Modifications may cause changes to dependent settings
- VAD**
- + Codec G723.1 Select
- + Codec T38 FAX Select
- + QoS

Figure 19 – Media Gateways Configuration Details Page.

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

The screenshot shows the 'Media Gateways Configuration Details Page'. The left sidebar contains a tree view with the following items:

- Zones
- Host and Route Tables
- Network Address Translation (NAT)
- QoS Thresholds
- Personal Directories
- Unicode Name Directory
- + Interfaces
- Engineered Values
- + Emergency Services
- + Geographic Redundancy
- + Software
- Customers
- Routes and Trunks
- Routes and Trunks
- D-Channels
- Digital Trunk Interface
- Dialing and Numbering Plans
- Electronic Switched Network
- Flexible Code Restriction
- Incoming Digit Translation
- Phones
- Templates
- Reports
- Properties
- Migration
- Tools
- + Backup and Restore
- Call Server Initialization
- Date and Time
- + Logs and reports
- Security
- + Passwords
- + Policies
- + Login Options

 The main configuration area includes:

- + Codec G729A (Select)
- + Codec G723.1 (Select)
- + Codec T38 FAX (Select)
- QoS
- Enable Nortel Automatic QoS
- Diffserv codepoint(DSCP) control packets: 40 (0 - 63)
- Diffserv codepoint(DSCP) voice packets: 46 (0 - 63)
- Call Server LAN
- Embedded LAN (ELAN) configuration
- Geographic redundancy
- Primary call server IP address: 47.248.100.147
- Primary call server hostname: Primary_CS
- Signaling port: 15000
- Broadcast port: 15001 (1024 - 65535)
- Telephony LAN (TLAN) configuration
- Signaling port: 5000
- Voice port: 5200 (1024 - 65535)
- Routes
- Buttons: Add, Remove, Save, Cancel, VGW Channels
- Text: Click Add to add routes to the IPMG
- Footer: * Mandatory fields of current configuration

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 another 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**

The screenshot shows the 'Nortel CS 1000 Element Manager' interface. The left sidebar contains the following items:

- UCM Network Services
- Home
- Links
- Virtual Terminals
- System
- + Alarms
- Maintenance
- + Core Equipment
- Peripheral Equipment
- IP Network
- Nodes: Servers, Media Cards
- Maintenance and Reports
- Media Gateways
- Zones
- Host and Route Tables
- Network Address Translation (NAT)
- QoS Thresholds

 The main area displays:

- Managing: 47.248.100.147 Username: admin
- System » IP Network » Zones
- Zones**
- Zones are used to group related information for either bandwidth or dial plan numbering purposes.
- Bandwidth Zones** (highlighted in a red box)
- Bandwidth zones are used for alternate routing of calls between IP stations and also used for bandwidth management.
- Numbering Zones**
- Numbering zones are used to route the calls through a centralized call server.

Figure 21 –Zones Page.

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

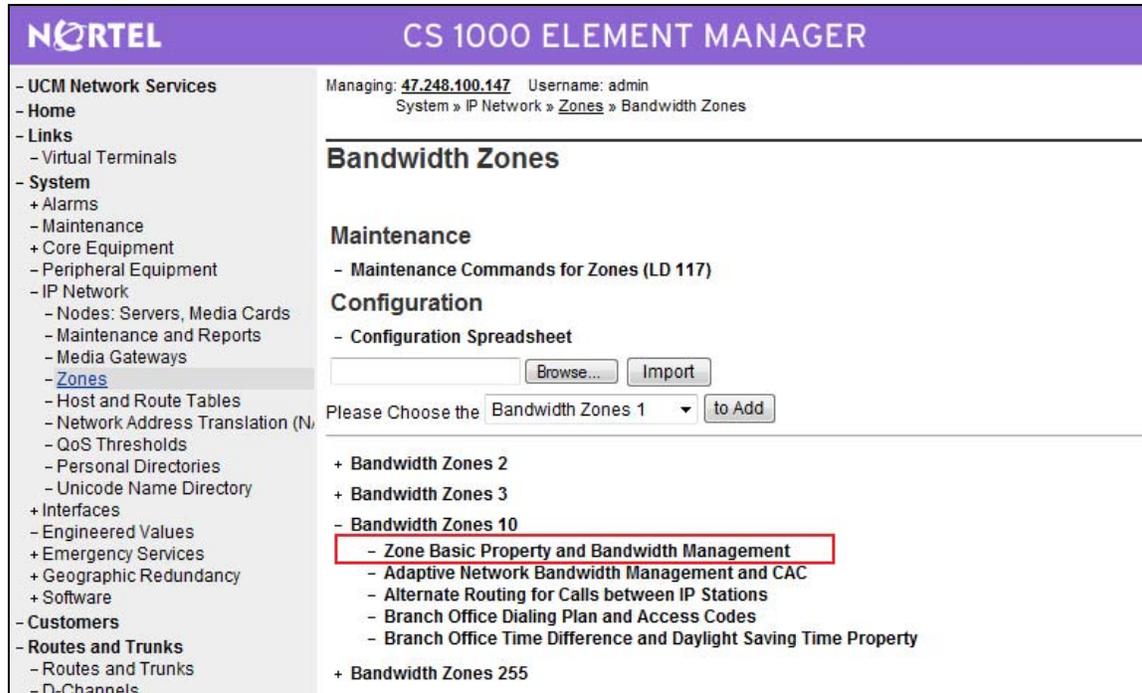


Figure 22 –Bandwidth Zones Page.

- 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, VGWetc
 - VTRK: is used for virtual trunk.

Managing: 47.248.100.147 Username: admin
System > IP Network > Zones > Bandwidth Zones > Bandwidth Zones 10 > Zone Basic Property and Bandwidth Management

Zone Basic Property and Bandwidth Management

Input Description	Input Value
Zone Number (ZONE):	10
Intrazone Bandwidth (INTRA_BW):	1000000
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	1000000
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	MO (MO)
Description (ZDES):	

Submit Refresh Delete Cancel

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**.

Managing: 47.248.100.147 Username: admin
System > IP Network > Zones > Bandwidth Zones > Bandwidth Zones 255 > Zone Basic Property and Bandwidth Management

Zone Basic Property and Bandwidth Management

Input Description	Input Value
Zone Number (ZONE):	255
Intrazone Bandwidth (INTRA_BW):	1000000
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	1000000
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	VTRK (VTRK)
Description (ZDES):	

Submit Refresh Delete Cancel

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**.

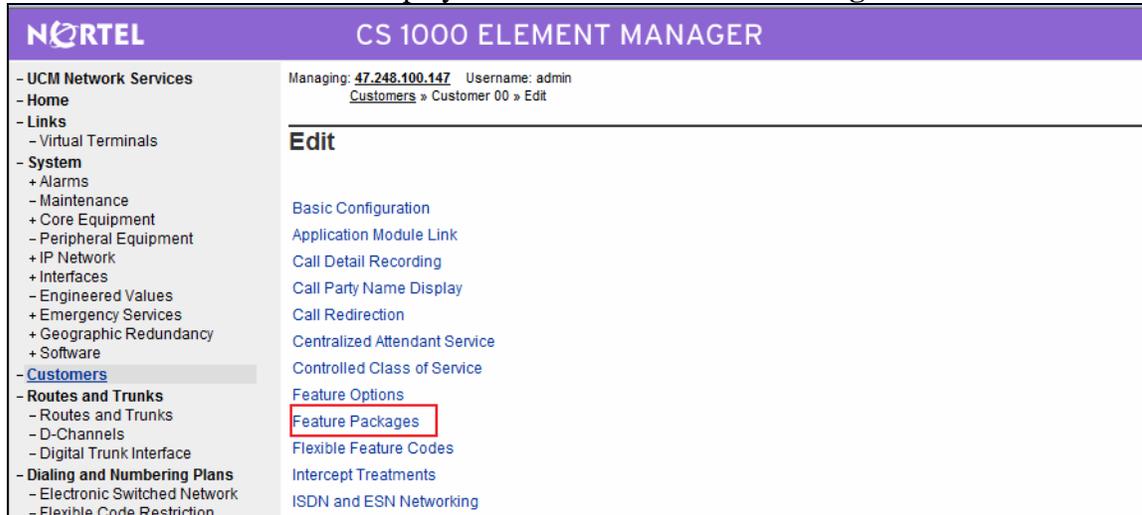


Figure 25 –Customer - feature packages Configuration Page.

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).

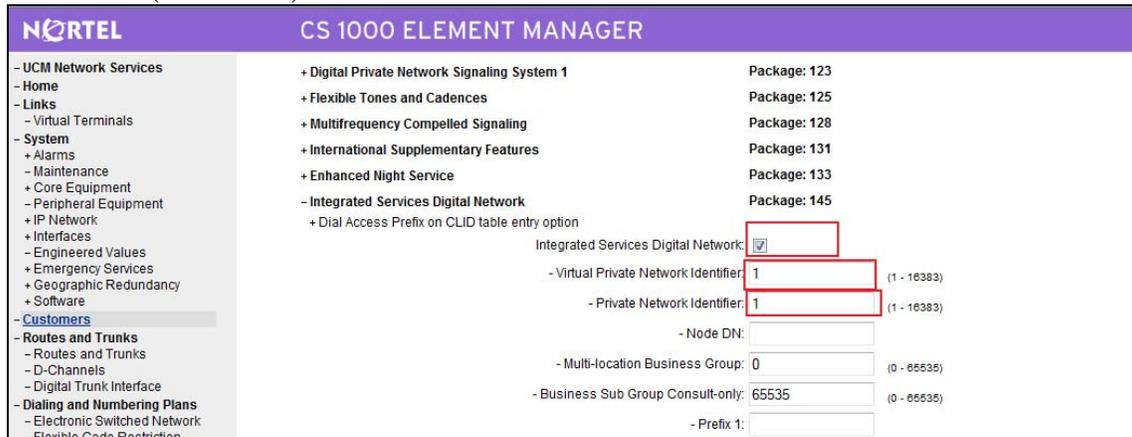


Figure 26 –Customer – ISDN Configuration Page.

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)**

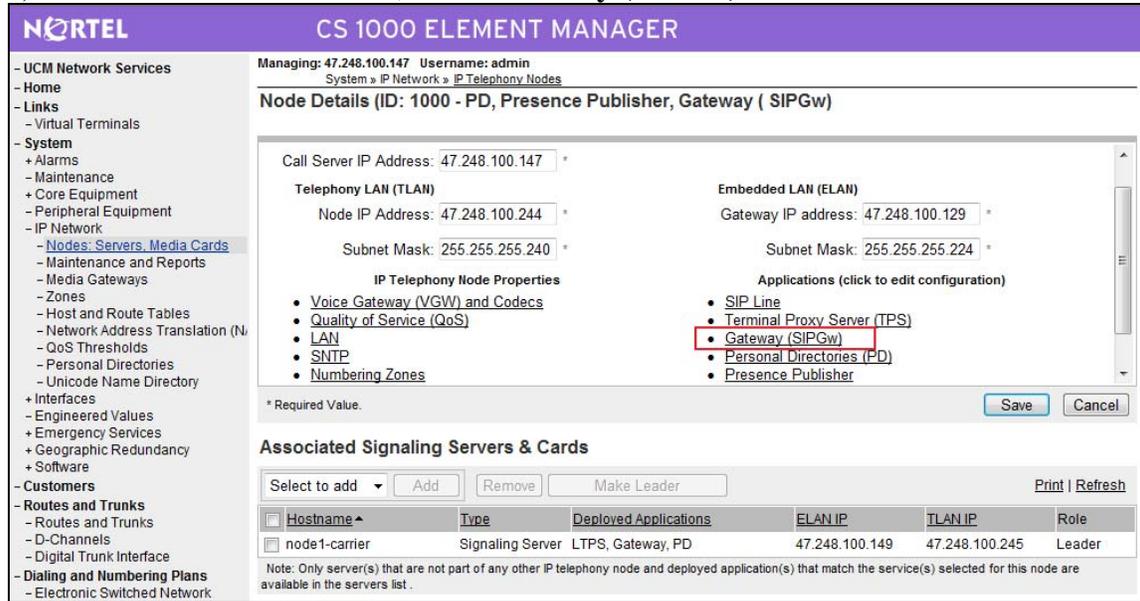


Figure 27 – Node Details – Gateway Configuration Page.

- 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: xxxxxxxxxxxxxxxx ---> 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

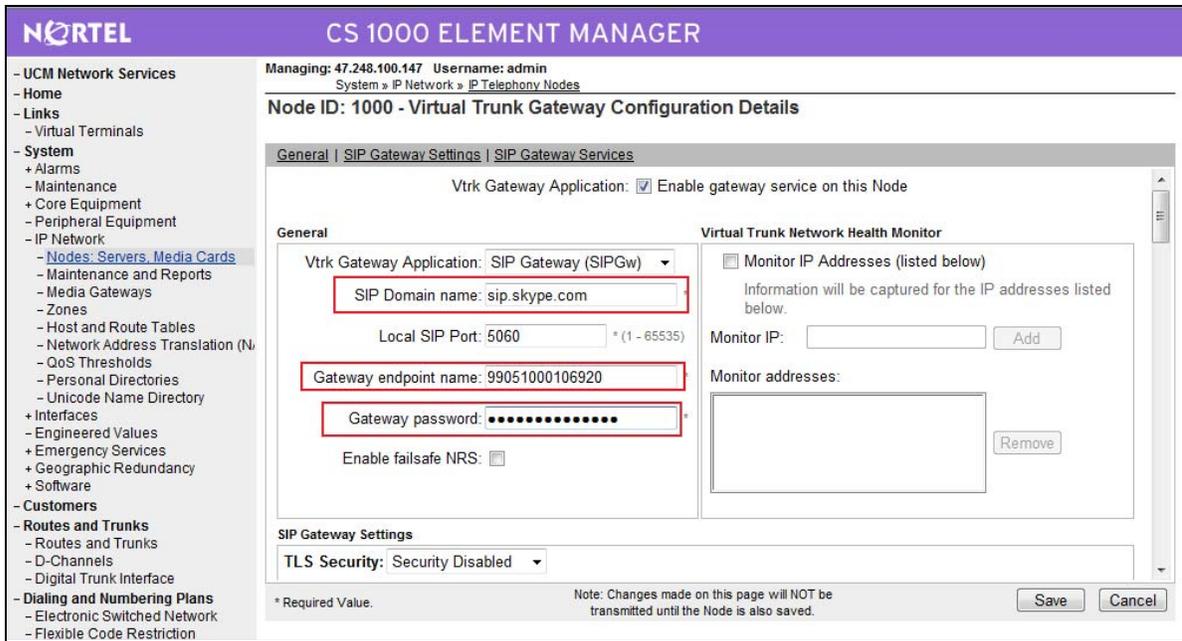


Figure 28 – Virtual Trunk Gateways Configuration Details Page.

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**

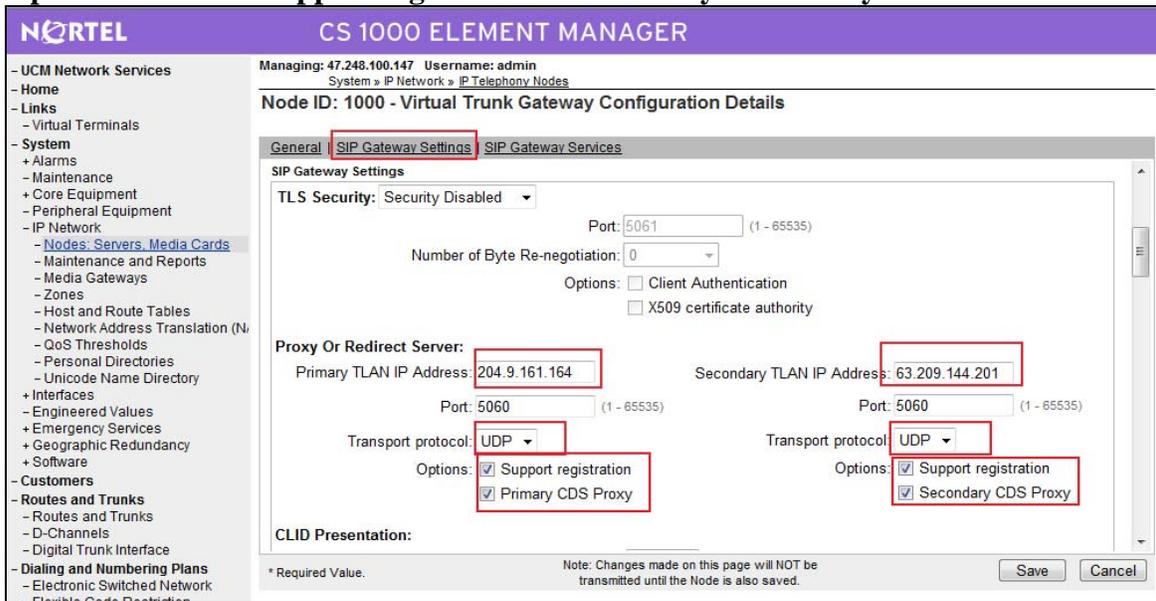


Figure 29 – Virtual Trunk Gateway Configuration Details Page.

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.

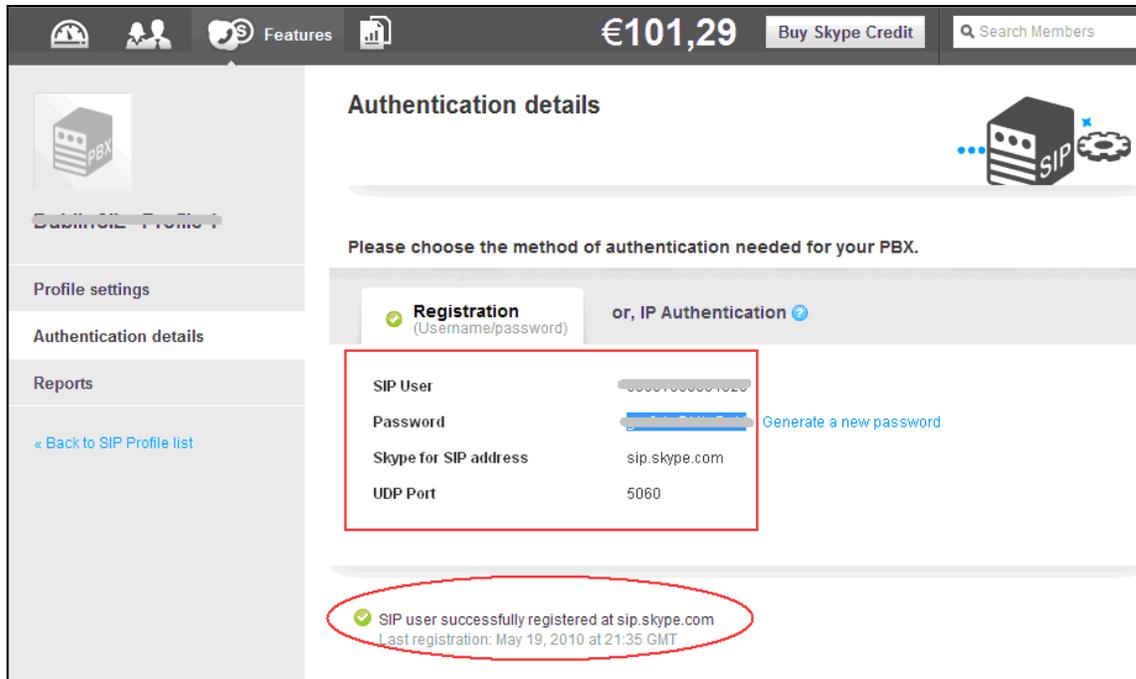


Figure 31 - CS1000 registered successfully to Skype.

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.

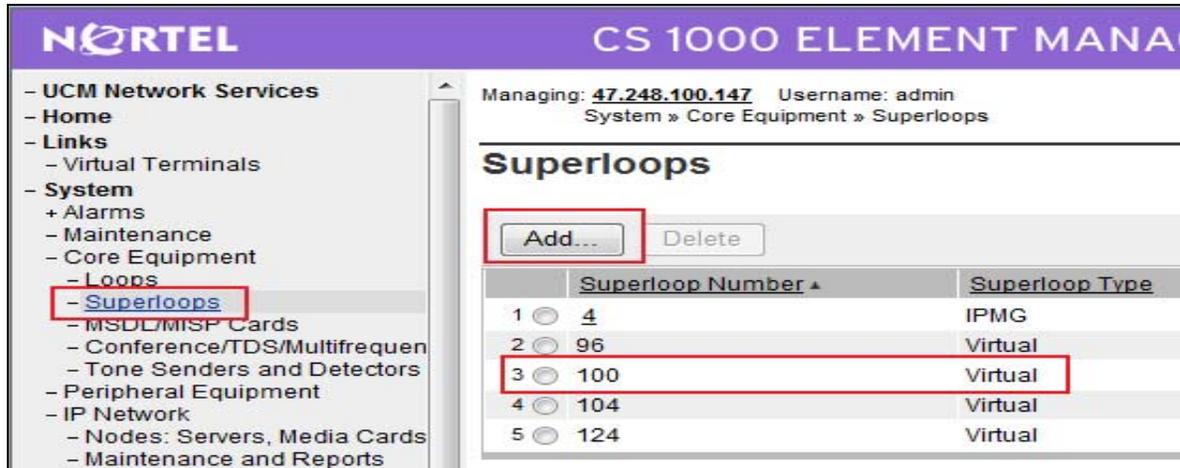


Figure 36 – Administer Virtual Super-Loop Page.

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**.

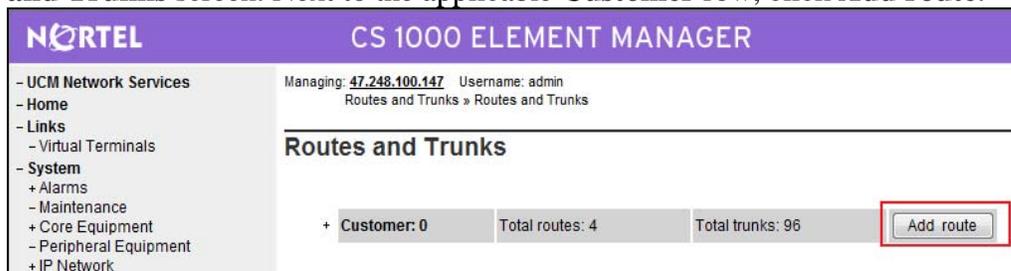


Figure 37 – Add route.

- 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.
 - o **Mode of operation (MODE):** Route uses ISDN Signaling Link (ISLD)
 - o **D channel number (DCH):** D-Channel number 100 (created in Section 4.5.3)
 - o **Network calling name allowed (NCNA):** Check the field.
 - o **Network call redirection (NCRD):** Check the field.
 - o **Insert ESN access code (INAC):** Check the field.

Customer 0, Route 100 Property Configuration

- Basic Configuration

Route data block (RDB) (TYPE)

Customer number (CUST)

Route number (ROUT)

Designator field for trunk (DES)

Trunk type (TKTP)

Incoming and outgoing trunk (ICOG)

Access code for the trunk route (ACOD)

Trunk type M911P (M911P)

The route is for a virtual trunk route (VTRK)

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

- Node ID of signaling server of this route (NODE) Range: 0 - 9999

- Protocol ID for the route (PCID)

- Print correlation ID in CDR for the route (CRID)

Integrated services digital network option (ISDN)

- Mode of operation (MODE)

- D channel number (DCH) Range: 0 - 254

- Interface type for route (IFC)

- Private network identifier (PNI) Range: 0 - 32700

- Network calling name allowed (NCNA)

- Network call redirection (NCRD)

-- Trunk route optimization (TRO)

- Recognition of DTI2 ABCD FALT signal for ISL (FALT)

- Channel type (CHTY)

- Call type for outgoing direct dialed TIE route (CTYP)

- Insert ESN access code (INAC)

Figure 38 – Route Configuration Details Page.

The screenshot shows the 'Basic Route Options' section of a configuration page. On the left is a navigation menu with categories like 'D-Channels', 'Dialing and Numbering Plans', 'Phones', 'Tools', and 'Security'. The main area contains several configuration options:

- Display of access prefix on CLID (DAPC)
- Attendant announcement (ATAN) No Attendant Announcement. (NO)
- Billing number required (BILN)
- Call detail recording (CDR)
- North American toll scheme (NATL) (highlighted with a red box)
- Controls or timers (CNTL)
- Conventional (Tie trunk only) (CNVT)
- Incoming DID digit conversion on this route (IDC) (highlighted with a red box)
 - Day IDC tree number (DCNO) 1 (Range: 0 - 254)
 - Night IDC tree number (NDNO) 1 (Range: 0 - 254)
- Display external dialed digits (DEXT)

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.

The screenshot shows the 'Routes and Trunks' page in the CS 1000 ELEMENT MANAGER. The page header includes the Nortel logo and 'CS 1000 ELEMENT MANAGER'. Below the header, it shows 'Managing: 47.248.100.147' and 'Username: admin'. The main content area is titled 'Routes and Trunks' and displays a summary table:

Customer	Total routes	Total trunks	Action
- Customer: 0	Total routes: 4	Total trunks: 96	<input type="button" value="Add route"/>
+ Route: 11	Type: TIE	Description: SIPL	<input type="button" value="Edit"/> <input type="button" value="Add trunk"/>
+ Route: 100	Type: TIE	Description: CARRIER	<input type="button" value="Edit"/> <input checked="" type="button" value="Add trunk"/> (highlighted with a red box)
+ Route: 101	Type: TIE	Description: ENTERPRISE	<input type="button" value="Edit"/> <input type="button" value="Add trunk"/>
- Route: 105	Type: DID	Description: 911	<input type="button" value="Edit"/> <input type="button" value="Add trunk"/>

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 (**STRO**): IMM
- Trunk Group Access Restriction (**TGAR**): Desired trunk group access restriction level
- Channel ID for this trunk (**CHID**): An available starting channel ID

Input Description	Input Value
Multiple trunk input number (MTINPUT)	32
Trunk data block (TYPE)	IP Trunk (IPTI)
Terminal Number (TN)	100 00 01 00
Designator field for trunk (DES)	Carrier
Extended Trunk (XTRK)	VTRK
Route number, Member number (RTMB)	100 1
Level 3 Signaling (SIGL)	
Card Density (CDEN)	
Start arrangement Incoming (STRI)	Immediate (IMM)
Start arrangement Outgoing (STRO)	Immediate (IMM)
Trunk Group Access Restriction (TGAR)	0
Channel ID for this trunk (CHID)	1
Increase or decrease the member numbers (INC)	Increase channel and member number (YES)
Class of Service (CLS)	Edit

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).

Managing: 47.248.100.147 Username: admin
Routes and Trunks » Routes and Trunks » Customer 0, Route 100, New Trunk Configuration » Class of Service Configuration

Class of Service Configuration

Input Description	Input Value
- ACD Priority (CLS)	ACD Priority not required (APN)
- Analog Semi-Permanent Connections (CLS)	Analog Semi-Permanent Connections Denied (SPCD)
- ARF Supervised COT (CLS)	
- Barring (CLS)	
- Battery Supervised COT (CLS)	
- Busy Tone Supervised COT (CLS)	
- Calling Line Identification (CLS)	
- Calling party (CLS)	Calling party Denied (CND)
- Central Office Ringback (CLS)	
- Centrex Switchhook Flash (CLS)	Centrex Switchhook Flash Denied (THFD)
- Dial Pulse (CLS)	Digitone (DTN)
- DTR PAD value (CLS)	
- Echo Canceling (CLS)	Echo Canceling Denied (ECD)
- Hong Kong DTI (CLS)	
- Loop Break Supervised COT (CLS)	
- Make-break ratio for dial pulse (CLS)	10 pulses per second (P10)
- Manual Incoming (CLS)	Manual Incoming Denied (MID)
- Media Security (CLS)	Media Security Never (MSNV)
- Network Hook Flash Over M911P (CLS)	

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)

- Log in UCM and EM (please refer to Section 4.1.1 for more detail)
- 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)**.

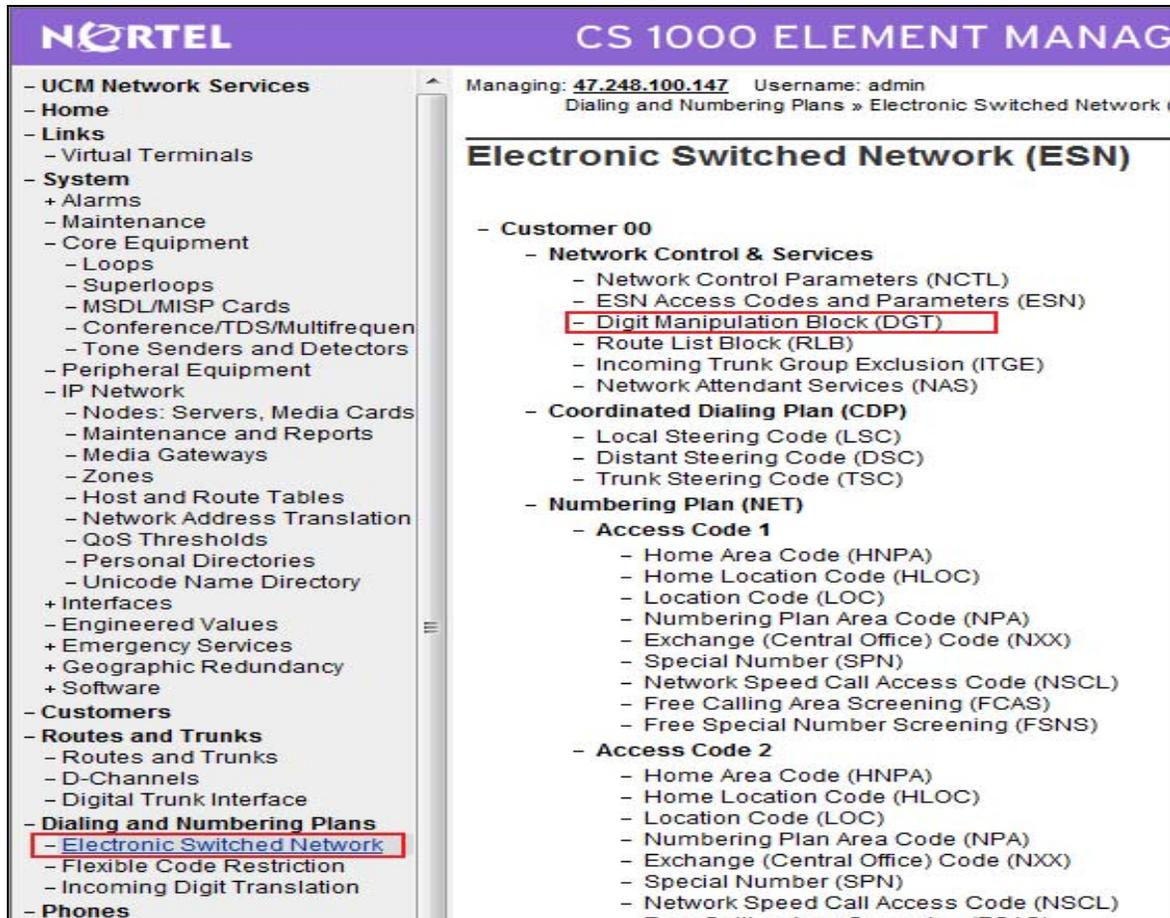


Figure 43 –ESN Configuration Details Page.

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**.

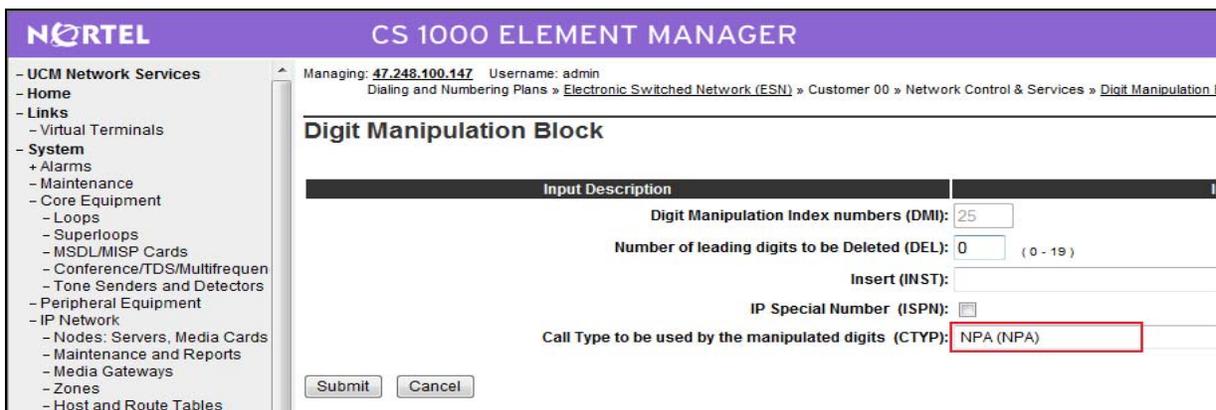


Figure 47 – DMI Configuration Details Page.

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)**.

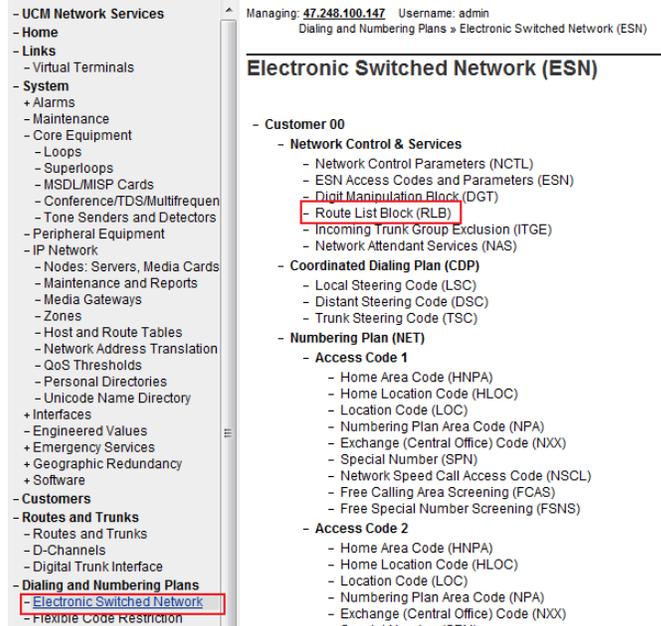


Figure 48 – ESN Configuration Details Page.

- 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)

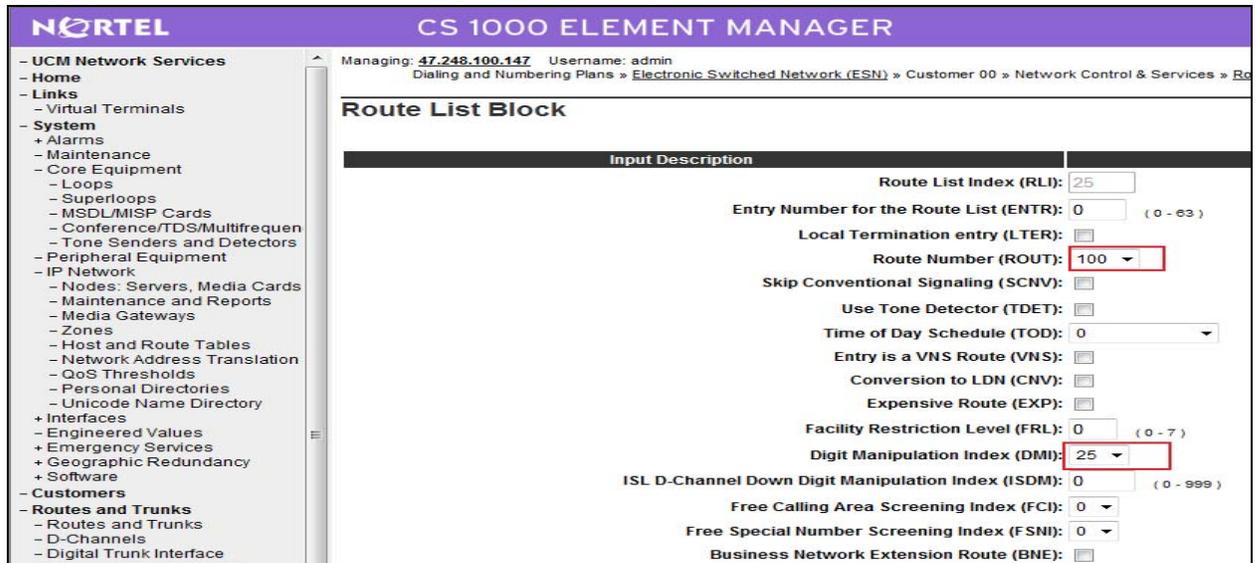


Figure 50 – Route List Blocks Configuration Details Page.

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

- Log in UCM and EM (please refer to Section 4.1.1 for more detail)
- 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**.
- Select an available value to and click to **Add** (in this case is 26)

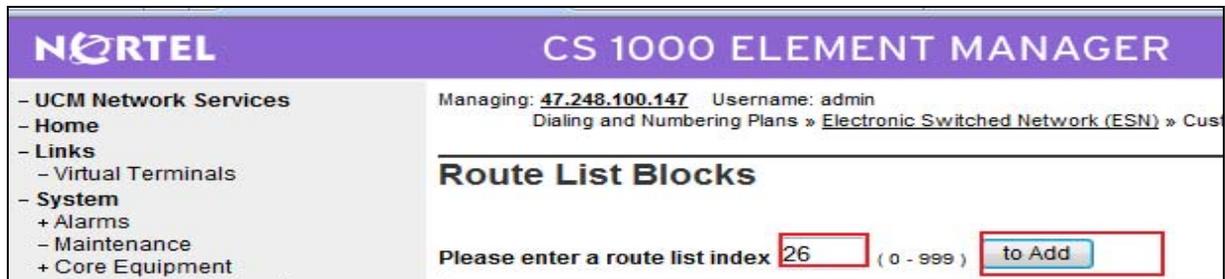


Figure 51 – Add a Route List Blocks.

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

Managing: 47.248.100.147 Username: admin
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Network Control & Services » Route

Route List Block

Input Description	Value
Route List Index (RLI):	26
Entry Number for the Route List (ENTR):	0 (0 - 99)
Local Termination entry (LTER):	<input type="checkbox"/>
Route Number (ROUT):	100
Skip Conventional Signaling (SCNV):	<input type="checkbox"/>
Use Tone Detector (TDET):	<input type="checkbox"/>
Time of Day Schedule (TOD):	0
Entry is a VNS Route (VNS):	<input type="checkbox"/>
Conversion to LDN (CNV):	<input type="checkbox"/>
Expensive Route (EXP):	<input type="checkbox"/>
Facility Restriction Level (FRL):	7 (0 - 7)
Digit Manipulation Index (DMI):	25
ISL D-Channel Down Digit Manipulation Index (ISDM):	0 (0 - 999)
Free Calling Area Screening Index (FCI):	0

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

- Log in UCM and EM (please refer to Section 4.1.1 for more detail)
- 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)**.

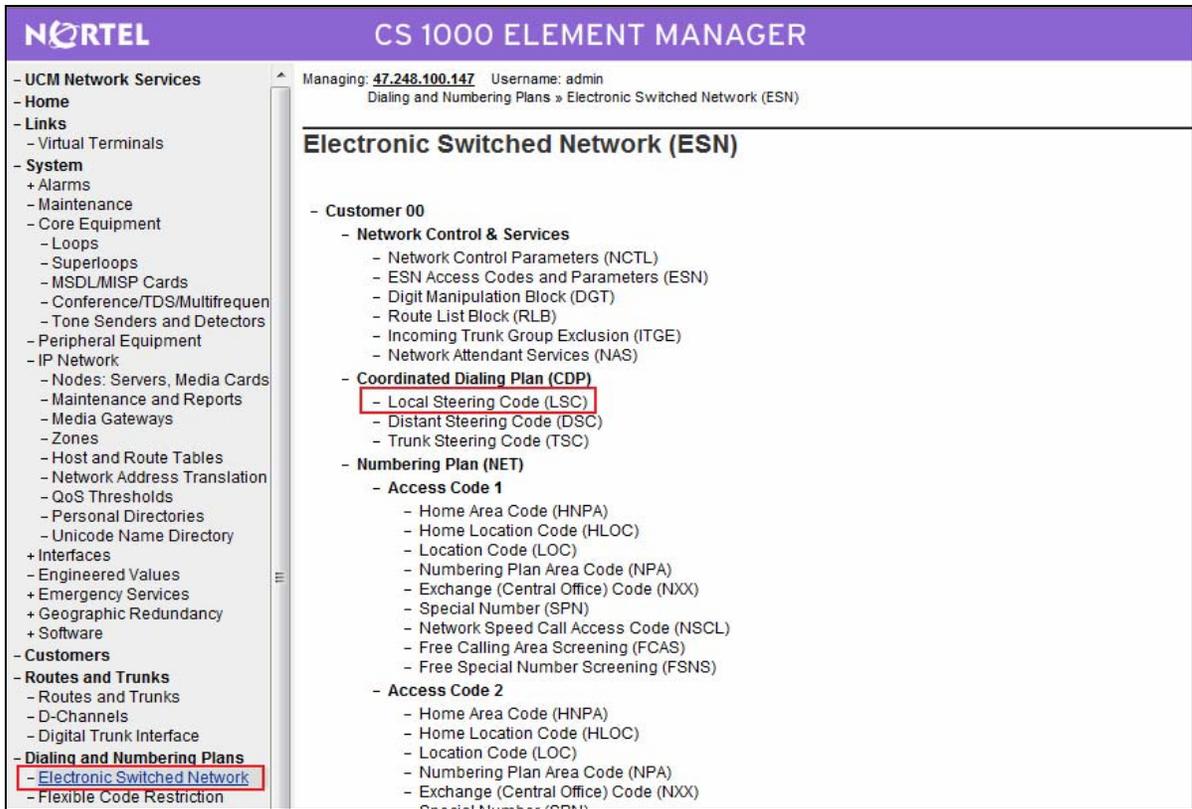


Figure 53 – ESN Configuration Page.

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**

The screenshot shows the 'Local Steering Code' configuration page in the Nortel CS 1000 Element Manager. The page has a purple header with the Nortel logo and 'CS 1000 ELEMENT MANAGER'. Below the header is a navigation menu on the left and a breadcrumb trail: 'Managing: 47.248.100.147 Username: admin' followed by 'Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Coordinated Dialing Plan (CDP) » Local Steering Code'. The main content area is titled 'Local Steering Code' and contains a table with the following fields:

Input Description	Value
Local Steering Code (LSC):	131
Digit Manipulation Index for LSC (DMI):	7
Number of digits to be deleted (DEL):	(1-7)

At the bottom of the form are 'Submit' and 'Cancel' buttons. The 'Submit' button is highlighted with a red box, and the '7' in the DMI dropdown is also highlighted with a red box.

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 Id 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 Id 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)**.

The screenshot displays the Nortel CS 1000 Element Manager interface. The top header shows 'NORTEL' and 'CS 1000 ELEMENT MANAGER'. Below the header, the user is logged in as 'admin' with the IP address '47.248.100.147'. The main content area is titled 'Electronic Switched Network (ESN)'. On the left, a navigation tree includes 'UCM Network Services', 'Home', 'Links', 'System', 'Customers', 'Routes and Trunks', 'Dialing and Numbering Plans', 'Phones', and 'Tools'. Under 'Dialing and Numbering Plans', 'Electronic Switched Network' is selected. The main content area shows the configuration for 'Customer 00', including 'Network Control & Services', 'Coordinated Dialing Plan (CDP)', and 'Numbering Plan (NET)'. The 'Special Number (SPN)' option is highlighted in red.

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)

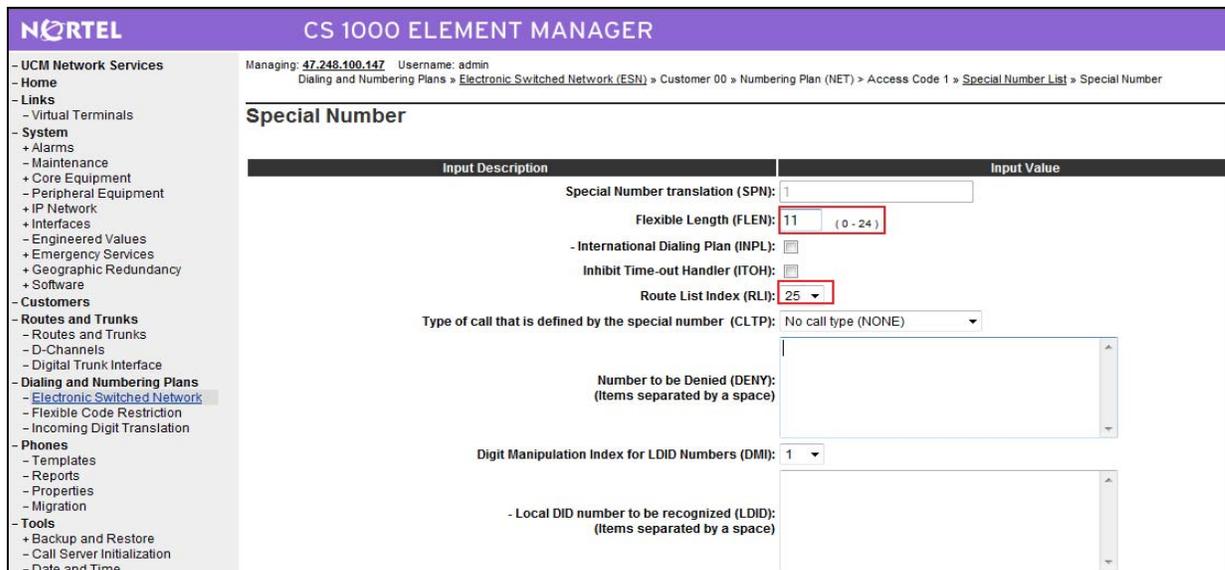


Figure 58 – SPN Configuration Details Page.

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.

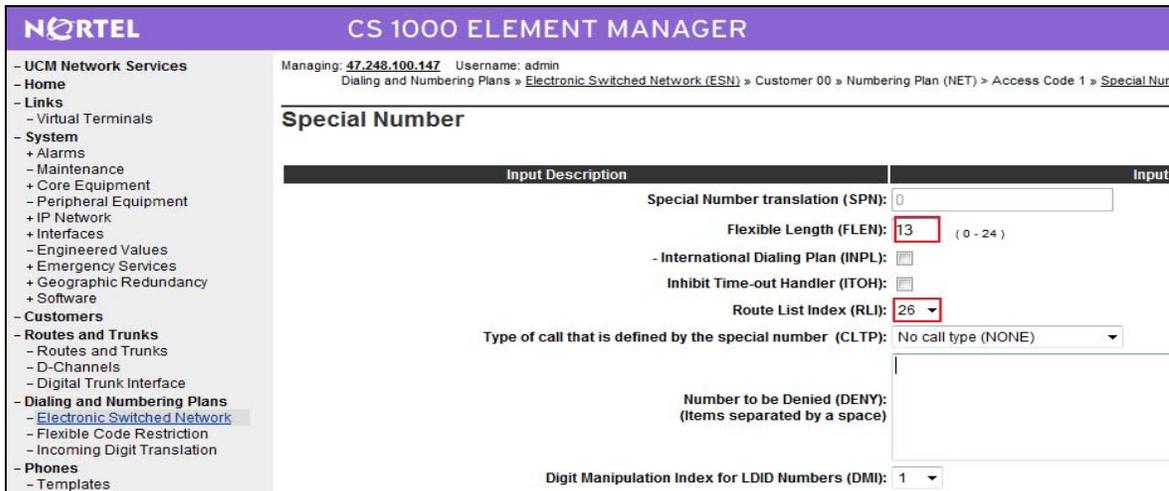


Figure 60 –SPN Configuration Details Page.

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 **fc**
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>

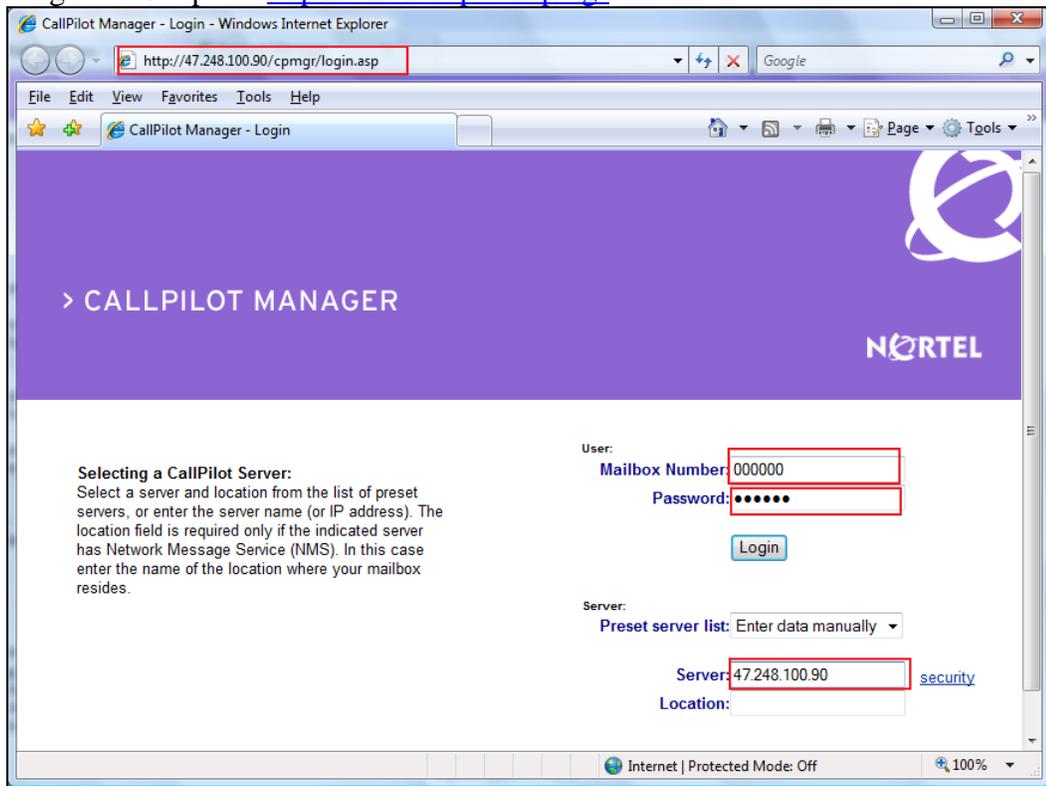


Figure 61 –Log in Callpilot.

b. **Select Configuration Wizard.**

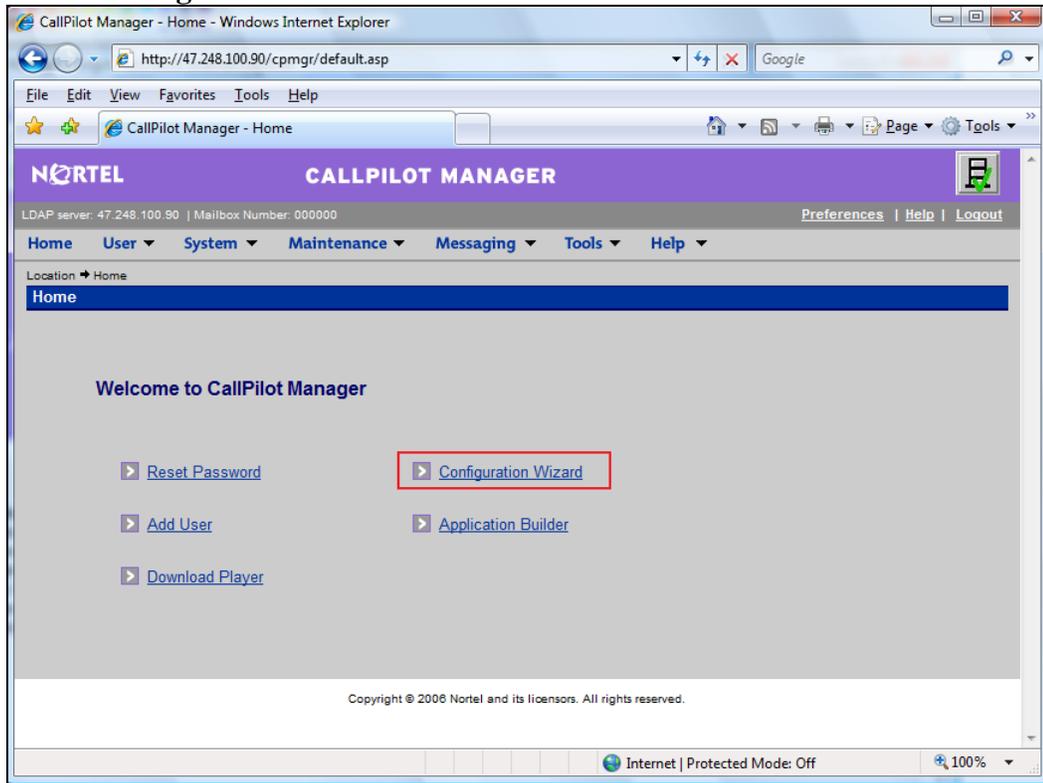


Figure 62 –Select Configuration Wizard.

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

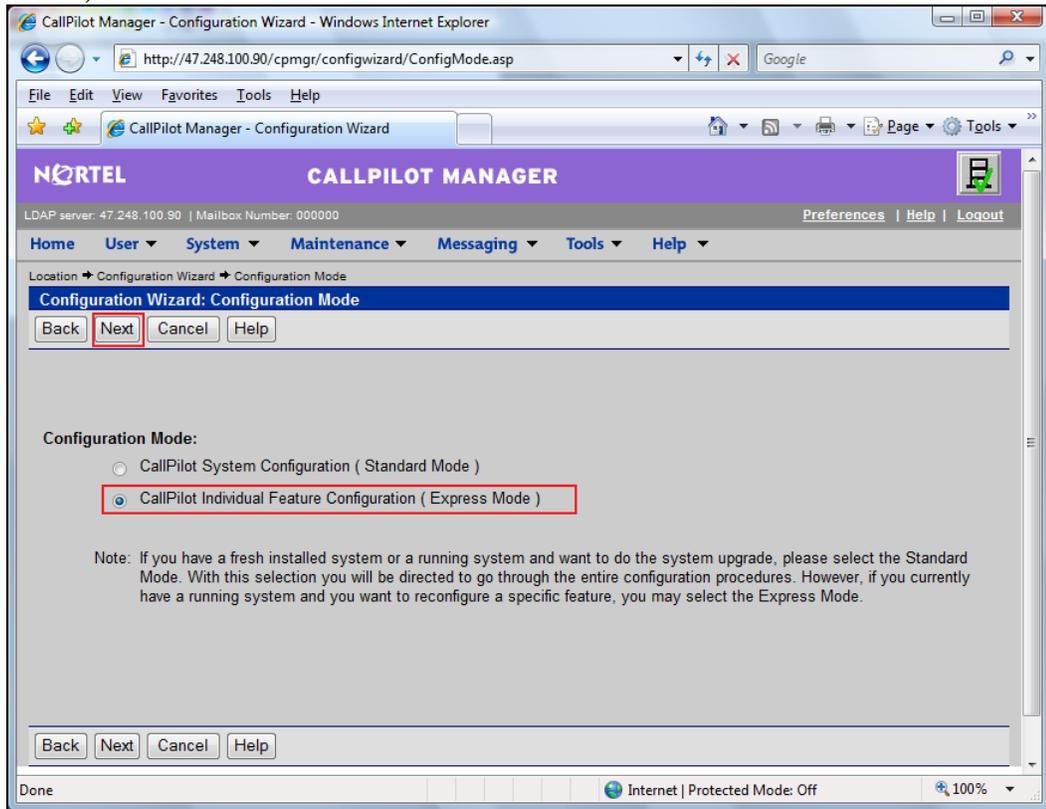


Figure 63 – Callpilot Configuration Details Page.

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

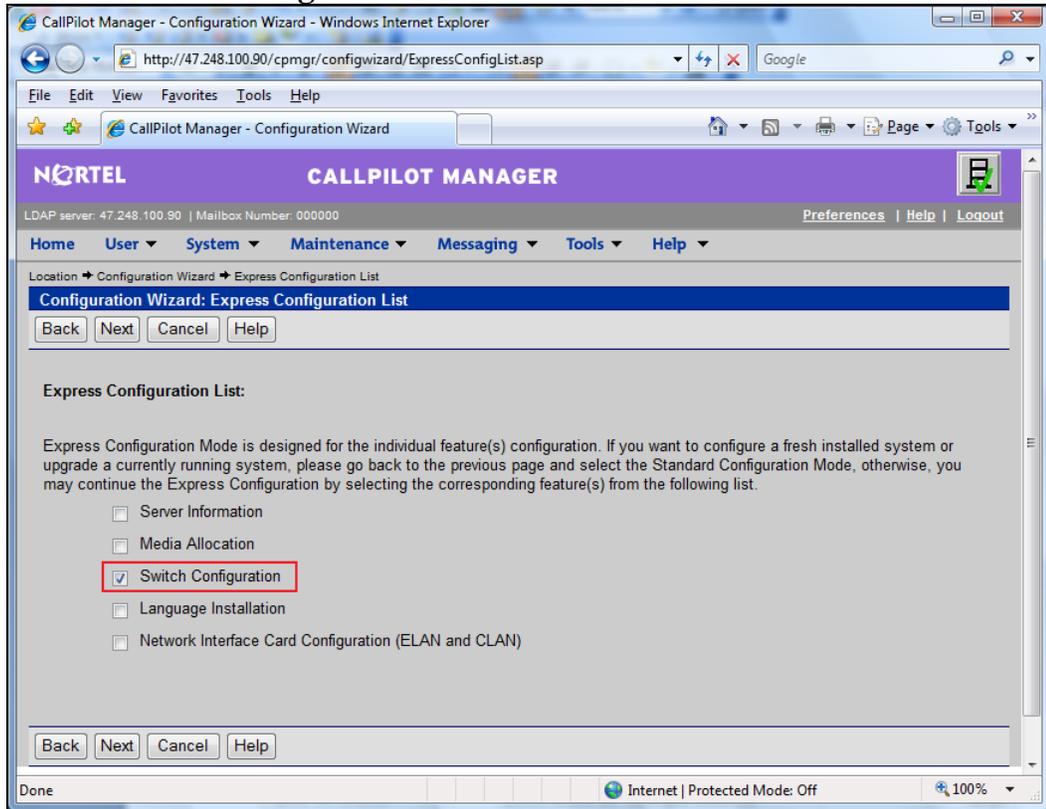


Figure 64 – Callpilot Configuration Details Page.

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

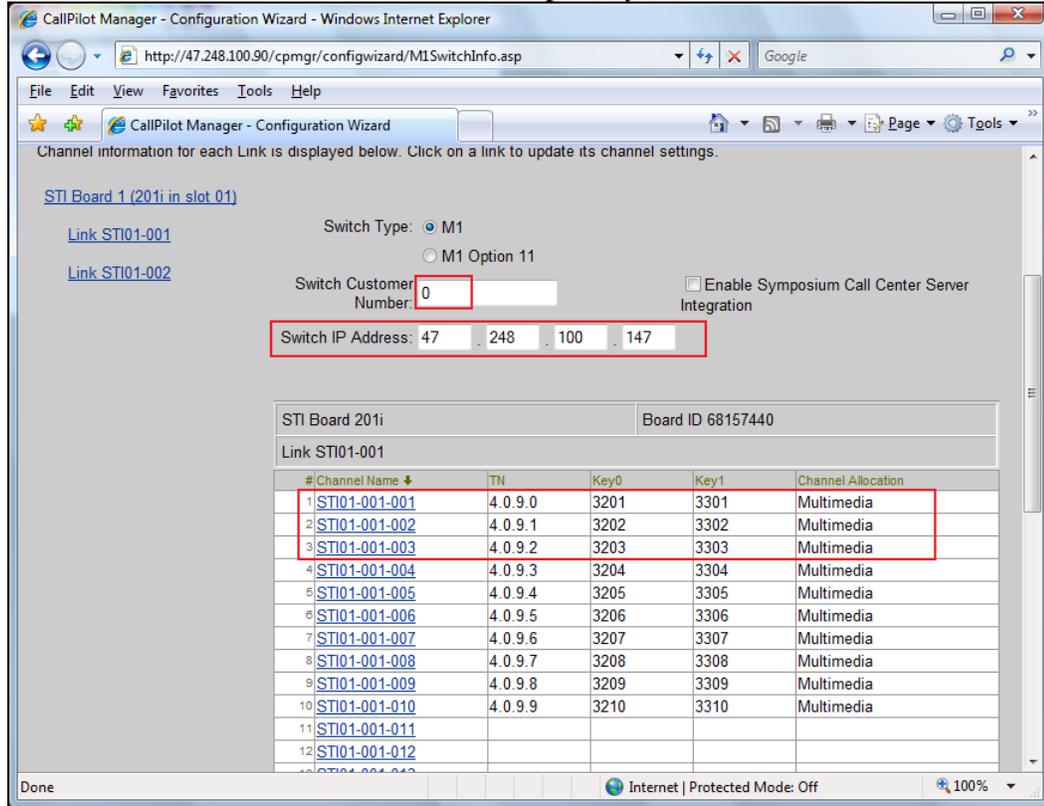


Figure 65 – Callpilot Configuration Details Page.

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**

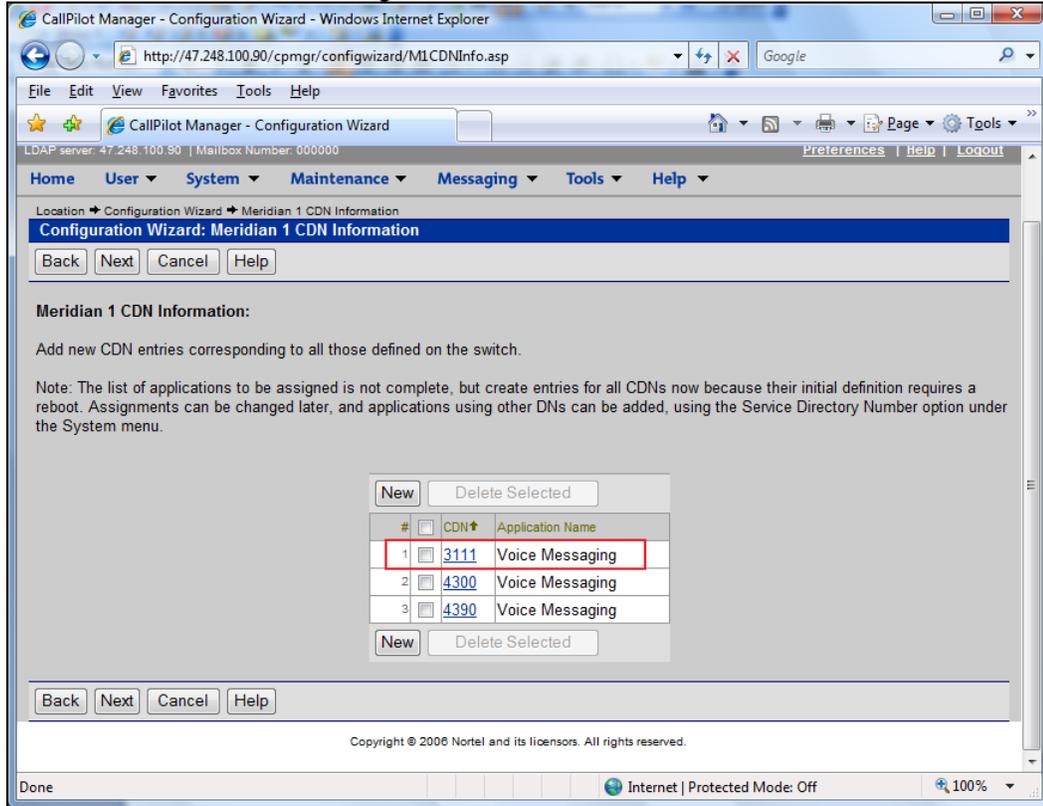


Figure 66 – Callpilot Configuration Details Page.

g. Click **Finish**.

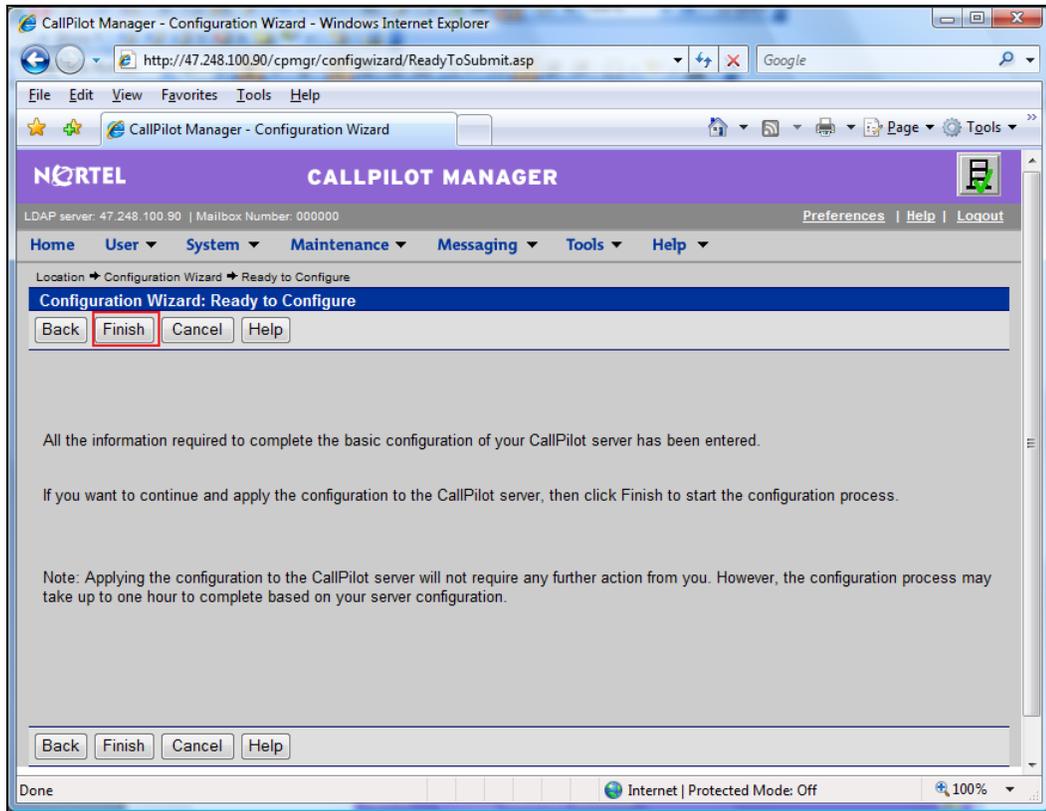


Figure 67 – Callpilot Configuration Details Page.

- 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 **4090**
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 RCB
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 HBT
DRDD EXR0
USMD USRD ULAD CCB
RTDD RBDD RBHD PGND OCB
FLXD FTTC DNDY DNO3
MCBN
FSD 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 Id 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.**

The screenshot shows the Nortel CS 1000 Element Manager interface. The left sidebar has 'System' and 'IP Network' highlighted. The main area displays 'IP Telephony Nodes' with a table of existing nodes. The 'Add...' button is highlighted in red. The table below shows the following data:

Node ID	Components	Enabled Applications	ELAN IP	TLAN IP	Status
1000	1	LTPS, PD, Presence Publisher, Gateway (SIPGw)	-	47.248.100.244	Synchronized
1001	1	SIP Line, LTPS, Gateway (SIPGw)	-	47.248.100.126	Synchronized
1002	1	SIP Li	-	47.248.100.120	Synchronized

Figure 69 – Add a new node for SIP line.

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

The screenshot shows the 'Node Details (ID: 1002 - SIP Li)' configuration page. The following fields are highlighted with red boxes:

- Node ID: 1002
- Call Server IP Address: 47.248.100.147
- Telephony LAN (TLAN) Node IP Address: 47.248.100.120
- Subnet Mask: 255.255.255.240
- Embedded LAN (ELAN) Gateway IP address: 47.248.100.129
- Subnet Mask: 255.255.255.224

The 'Applications' section shows 'SIP Line' selected. Below, the 'Associated Signaling Servers & Cards' table is visible:

Hostname	Type	Deployed Applications	ELAN IP	TLAN IP	Role
sip1-6	Signaling Server	SIP Line	47.248.100.131	47.248.100.119	Leader

Figure 70 –Configure SIP line.

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

NORTEL CS 1000 ELEMENT MANAGER
 Managing: 47.248.100.147 Username: admin
 System » IP Network » IP Telephony Nodes
 Node ID: 1002 - SIP Line Configuration Details

General | SIP Line Gateway Settings | SIP Line Gateway Service

SIP Line Gateway Application: Enable gateway service on this Node

General

SIP Domain name: *

SLG endpoint name:

SLG Group ID:

SLG Local Sip Port: (1 - 65535)

SLG Local Tls Port: (1 - 65535)

Virtual Trunk Network Health Monitor

Monitor IP Addresses (listed below)
 Information will be captured for the IP addresses listed below.

Monitor IP:

Monitor addresses:

SIP Line Gateway Settings

Security Policy: Security Disabled

Number of Byte Re-negotiation:

Options: Client Authentication
 x509 Certificate Authentication Enabled

* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved.

Figure 71 – SIP-Line Configuration Details Page.

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

NORTEL CS 1000 ELEMENT MANAGER
 Managing: 47.248.100.147 Username: admin
 System » IP Network » IP Telephony Nodes
 Node ID: 1002 - SIP Line Configuration Details

General | SIP Line Gateway Settings | SIP Line Gateway Service

x509 Certificate Authentication Enabled

SIP Line Gateway Service

Branch / GR Office Settings:

SLG Role:

SLG Mode:

MO SLG IP:

MO SLG Port: (1 - 65535)

MO SLG Transport:

GR SLG IP:

GR SLG Port: (1 - 65535)

GR SLG Transport:

IVR Settings:

SLG IVR Proxy IP:

SLG IVR Proxy Port: (1 - 65535)

* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved.

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
SIPLD 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
OVLN 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

>*1d 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

SIPL **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**

Nortel Softphone 3456 Login

Username: nhan

Password: *****

Remember name

Remember password

Sign in automatically

[Forgot your password?](#)

Server:

Skip Sign in

Figure 74 – SMC3456 Log In (Con.)

c) The SMC3456 Client will be displayed as follows.

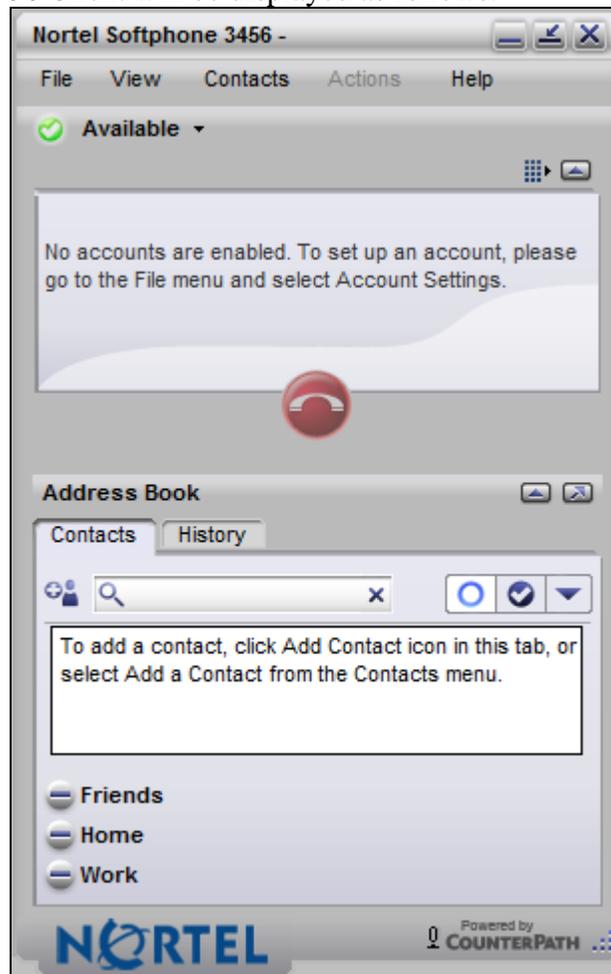


Figure 75 - SMC3456 Client

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**.

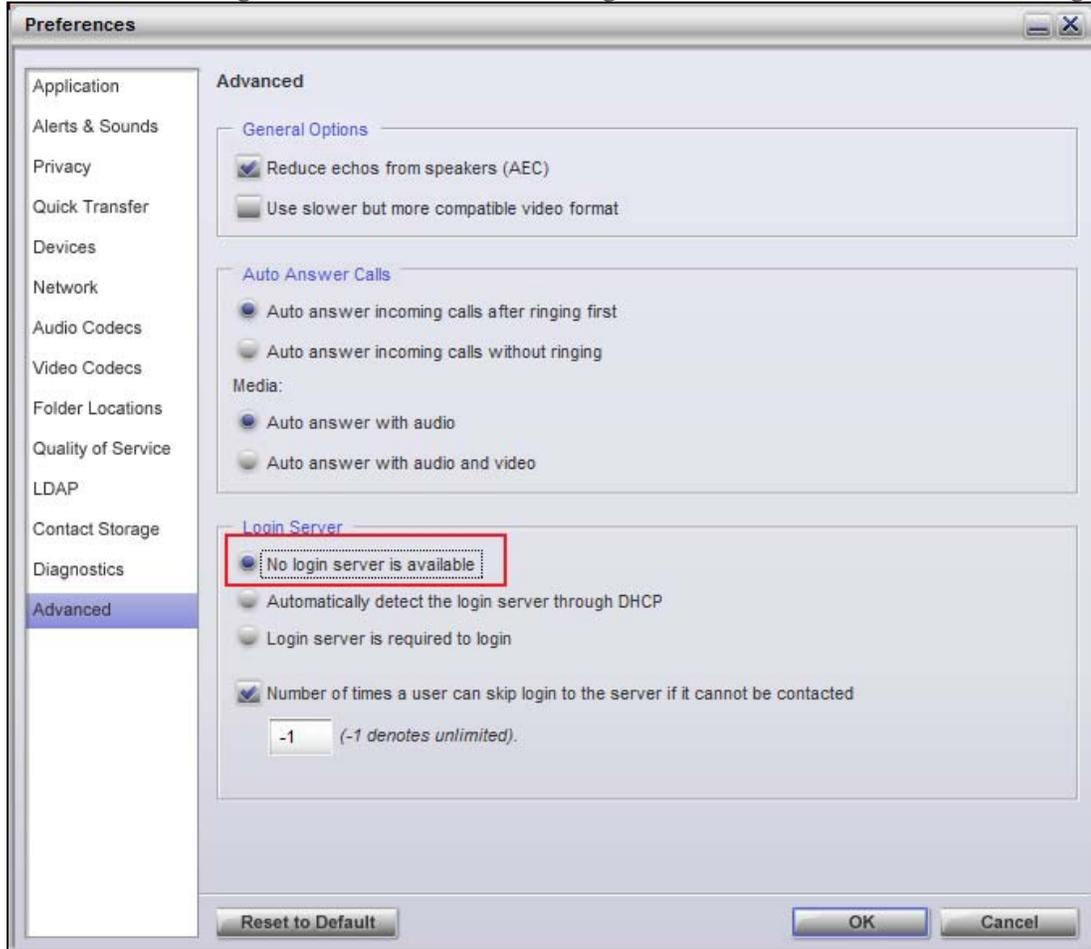


Figure 76 – Advanced Options Menu

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**.

The screenshot shows a window titled "SIP Account" with a close button (X) in the top right corner. The window contains several tabs: "Account", "Voicemail", "Topology", "Presence", "Storage", "Security", and "Advanced". The "Account" tab is selected. The "Account" section includes fields for "Account name" (4197) and "Protocol" (SIP). Below this is a "User Details" section with fields for "User ID" (4197@interop.com), "Password" (****), "Display name" (4197), and "Authorization name" (4197). A note next to the User ID field reads "e.g. joseph@domain.com". The "Domain Proxy" section has a checked checkbox for "Register with domain and receive calls". Under "Send outbound via:", the "Proxy" radio button is selected, with an "Address" field containing "47.248.100.120:5070". The "Domain" radio button is unselected. At the bottom, there is a "Dial plan" field containing "#2\|a\|a.T;match=1;prestrip=2;". At the bottom right of the window are "OK" and "Cancel" buttons.

Figure 78 – SIP Account Detail Settings

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

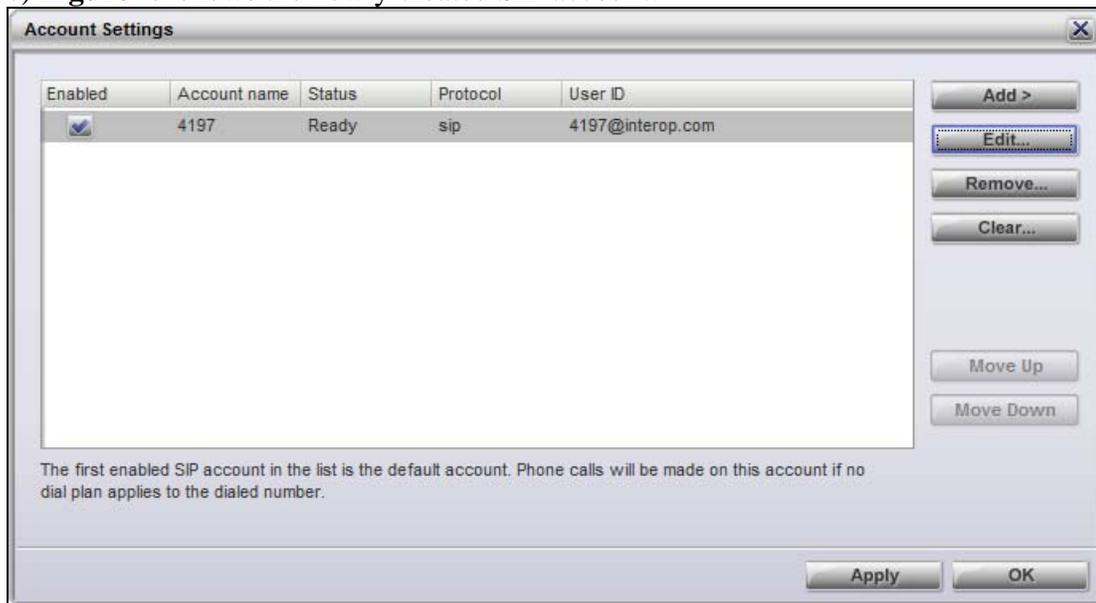


Figure 79 – New Created SIP Account

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)

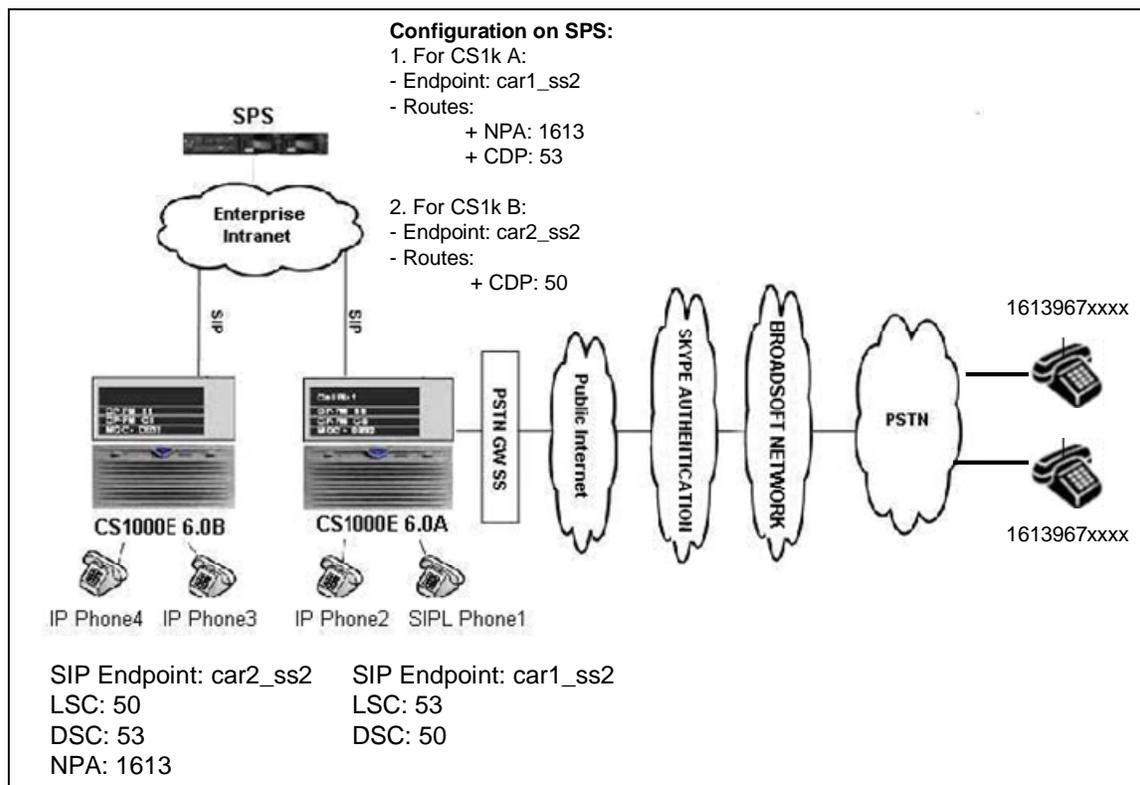


Figure 80 – Network Topology for Multi-system (tandem calls)

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
OVLN 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

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d. Administer SIP trunk gateway to SPS.

The screenshot displays the 'Virtual Trunk Gateway Configuration Details' for Node ID 1001. The interface includes a left-hand navigation menu with categories like 'UCM Network Services', 'System', 'Interfaces', 'Customers', 'Routes and Trunks', and 'Dialing and Numbering Plans'. The main configuration area is divided into several sections:

- General:**
 - Vtrk Gateway Application: Enable gateway service on this Node
 - Vtrk Gateway Application: SIP Gateway (SIPGw)
 - SIP Domain name: interop.com
 - Local SIP Port: 5060
 - Gateway endpoint name: car1_ss2
 - Gateway password: [Redacted]
 - Enable failsafe NRS:
- Virtual Trunk Network Health Monitor:**
 - Monitor IP Addresses (listed below)
 - Information will be captured for the IP addresses listed below.
 - Monitor IP: [Input field] [Add]
 - Monitor addresses: [List area] [Remove]
- SIP Gateway Settings:**
 - TLS Security: Security Disabled

At the bottom, there is a note: 'Note: Changes made on this page will NOT be transmitted until the Node is also saved.' and buttons for 'Save' and 'Cancel'.

Figure 81 – Virtual trunk gateway Configuration Details.

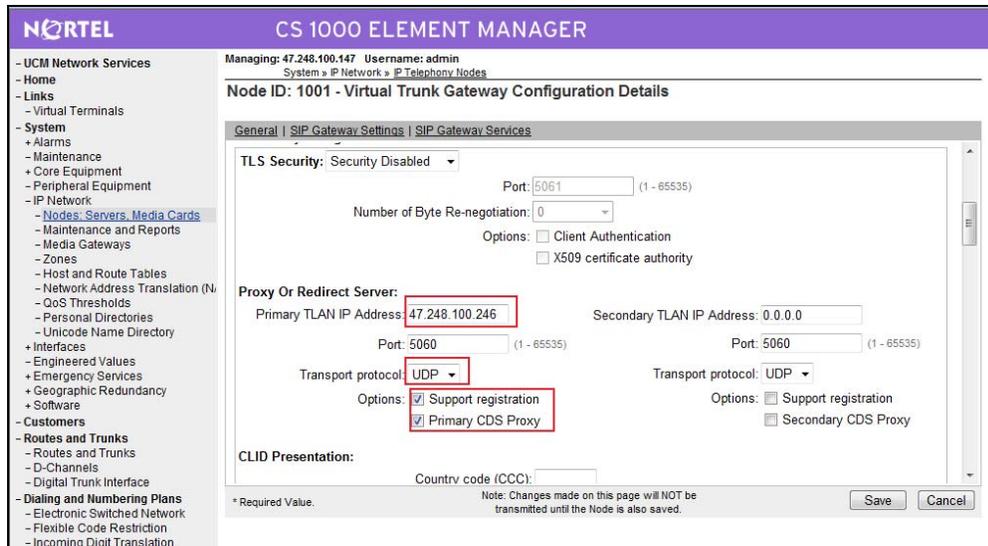


Figure 82 – Virtual Trunk Gateway Configuration Details.

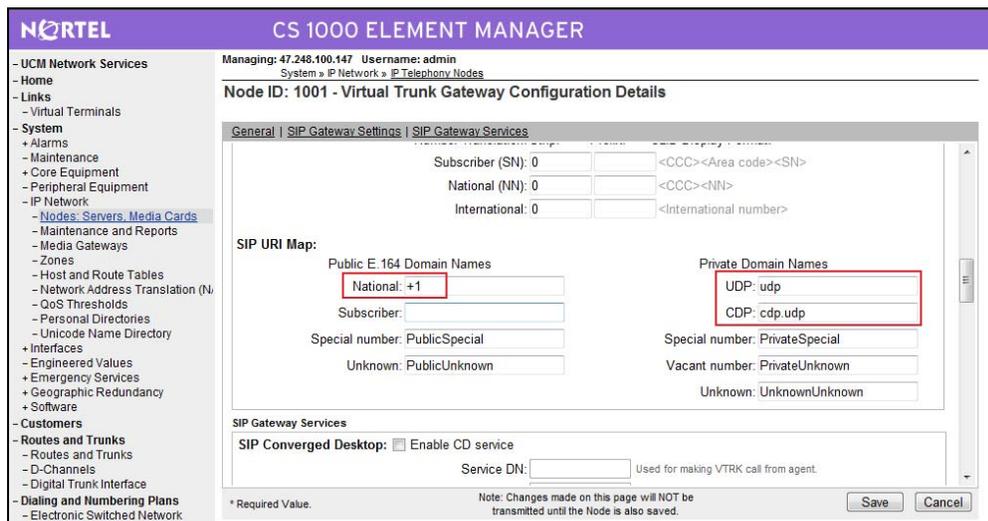


Figure 83 – Virtual trunk gateway Configuration Details.

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

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

Managing: 47.248.100.147 Username: admin
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Network Control & Services » Digit Manipulation Block List » Digit Manipulation Block

Digit Manipulation Block

Input Description	Input Value
Digit Manipulation Index numbers (DMI):	50
Number of leading digits to be Deleted (DEL):	0 (0 - 19)
Insert (INST):	
IP Special Number (ISPNUM):	
Call Type to be used by the manipulated digits (CTYP):	Coordinated Dialing Plan (CDP)

Buttons: Submit, Refresh, Delete, Cancel

Figure 84 – DMI Configuration Details.

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

Managing: 47.248.100.147 Username: admin
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Network Control & Services » Route List Blocks » Route List Block » Data Entry of a Route List Block

Data Entry of a Route List Block

Route List Block Index: 50

Input Description	Input Value
Entry Number for the Route List (ENTR):	0
Local Termination entry (LTER):	
Route Number (ROUT):	101
Skip Conventional Signaling (SCNV):	
Use Tone Detector (TDTE):	
Time of Day Schedule (TOD):	0
Entry is a VNS Route (VNS):	
Conversion to LDN (CHV):	
Expensive Route (EXP):	
Facility Restriction Level (FRL):	0 (0 - 7)
Digit Manipulation Index (DMI):	50
ISL D-Channel Down Digit Manipulation Index (ISDM):	0 (0 - 999)
Free Calling Area Screening Index (FCI):	0
Free Special Number Screening Index (FSNI):	
Business Network Extension Route (BNE):	
Strategy on Congestion (SBOC):	No Reroute (NRR)
-QSIG Alternate Routing Causes (COPT):	QSIG Alternate Routing Cause 1
ISDN Drop Back Busy (IDBB):	Drop Back Disabled (DBD)
ISDN Off-Hook Queuing Option (IOHQ):	
Off-Hook Queuing Allowed (OHQ):	
Call Back Queuing Allowed (CBQ):	

Buttons: Submit, Refresh, Delete, Cancel

Figure 85 – Route List Blocks Configuration Details.

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

Managing: 47.248.100.147 Username: admin
Dialing and Numbering Plans » Electronic Switched Network (ESN) » Customer 00 » Coordinated Dialing Plan (CDP) » Distant Steering Code List » Distant Steering Code

Distant Steering Code

Input Description	Input Value
Distant Steering Code (DSC):	50
Flexible Length number of digits (FLEN):	6 (0 - 10)
Display (DSP):	Local Steering Code (LSC)
Remote Radio Paging Access (RRPA):	
Route List to be accessed for trunk steering code (RLI):	50
Collect Call Blocking (CCBA):	
maximum 7 digit NPA code allowed (NPA):	
maximum 7 digit NXX code allowed (NXX):	

Buttons: Submit, Refresh, Delete, Cancel

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**.

The screenshot shows the 'Digit Manipulation Block' configuration page in the CS 1000 ELEMENT MANAGER. The page title is 'CS 1000 ELEMENT MANAGER' and the user is 'admin'. The breadcrumb trail is 'Dialing and Numbering Plans > Electronic Switched Network (ESN) > Customer 00 > Network Control & Services > Digit Manipulation Block List > Digit Manipulation Block'. The main content area is titled 'Digit Manipulation Block' and contains a table with two columns: 'Input Description' and 'Input Value'. The table has the following rows:

Input Description	Input Value
Digit Manipulation Index numbers (DMI):	2
Number of leading digits to be Deleted (DEL):	2 (0 - 19)
Insert (INST):	
IP Special Number (ISP):	<input type="checkbox"/>
Call Type to be used by the manipulated digits (CTYP):	Coordinated Dialing Plan (CDP)

At the bottom of the form are buttons for 'Submit', 'Refresh', 'Delete', and 'Cancel'.

Figure 87 – DMI Configuration Details.

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

The screenshot shows the 'Local Steering Code' configuration page in the CS 1000 ELEMENT MANAGER. The page title is 'CS 1000 ELEMENT MANAGER' and the user is 'admin'. The breadcrumb trail is 'Dialing and Numbering Plans > Electronic Switched Network (ESN) > Customer 00 > Coordinated Dialing Plan (CDP) > Local Steering Code'. The main content area is titled 'Local Steering Code' and contains a table with two columns: 'Input Description' and 'Input Value'. The table has the following rows:

Input Description	Input Value
Local Steering Code (LSC):	53
Digit Manipulation Index for LSC (DMI):	2
Number of digits to be deleted (DEL):	(1 - 7)

At the bottom of the form are buttons for 'Submit', 'Refresh', 'Delete', and 'Cancel'.

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 Id 15. This section prints FCR configuration details.

```
>Id 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 Id 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**.

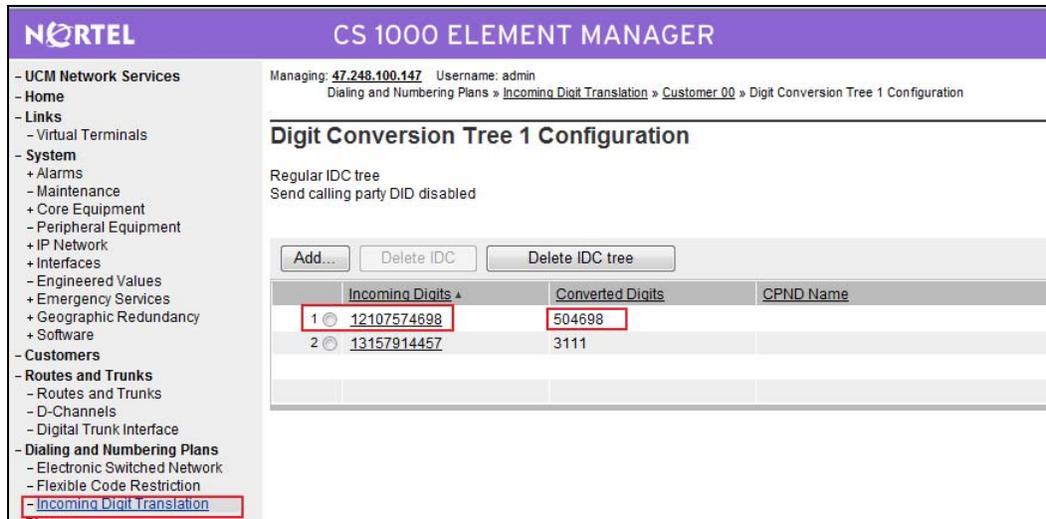


Figure 89 – Digit Conversion Tree 1 Configuration Details.

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
OVLN 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

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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.

NORTEL CS 1000 ELEMENT MANAGER

Managing: 47.248.100.138 Username: admin
System » IP Network » IP Telephony Nodes

Node ID: 2001 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services

General

Vtrk Gateway Application: SIP Gateway (SIPGw)
 SIP Domain name: interop.com
 Local SIP Port: 5060 * (1 - 65535)
 Gateway endpoint name: car2_ss2
 Gateway password: *
 Enable failsafe NRS:

virtual trunk network health monitor

Monitor IP Addresses (listed below)
 Information will be captured for the IP addresses listed below.
 Monitor IP: Add
 Monitor addresses: Remove

SIP Gateway Settings

TLS Security: Security Disabled
 Port: 5061 (1 - 65535)
 Number of Byte Re-negotiation: 0

* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved. Save Cancel

Figure 90 – Virtual Trunk Gateway Configuration Details.

NORTEL CS 1000 ELEMENT MANAGER

Managing: 47.248.100.138 Username: admin
System » IP Network » IP Telephony Nodes

Node ID: 2001 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services

Number of Byte Re-negotiation: 0
 Options: Client Authentication
 X509 certificate authority

Proxy Or Redirect Server:

Primary TLAN IP Address: 47.248.100.246
 Secondary TLAN IP Address: 0.0.0.0
 Port: 5060 (1 - 65535) Port: 5060 (1 - 65535)
 Transport protocol: UDP Transport protocol: TCP
 Options: Support registration
 Primary CDS Proxy Secondary CDS Proxy

CLID Presentation:

Country code (CCC):
 Area code: NPA in North America

* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved. Save Cancel

Figure 91 – Virtual trunk gateway configuration details

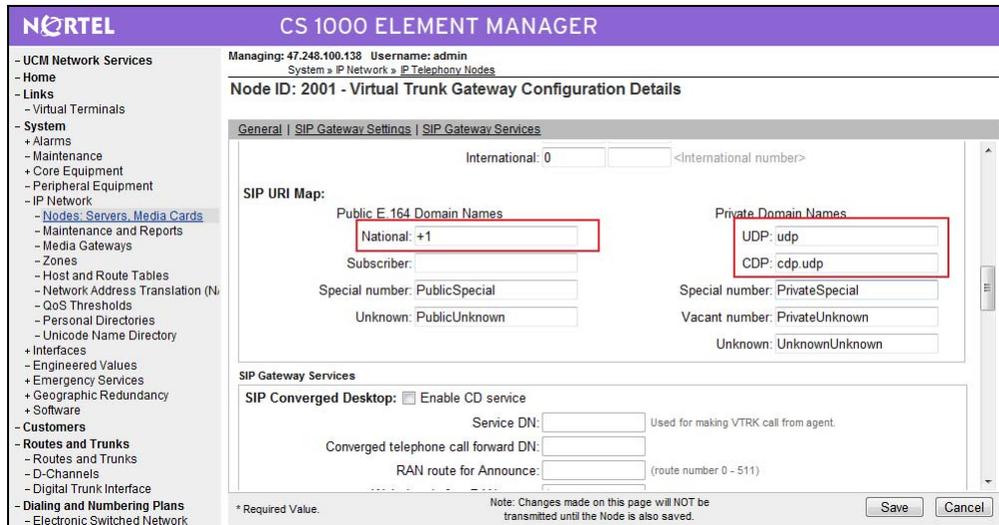


Figure 92 – Virtual Trunk Gateway Configuration Details.

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

The screenshot shows the Nortel Unified Communications Management (UCM) interface. The left sidebar contains a navigation menu with categories like Network, User Services, Security, and Tools. The main content area is titled 'UNIFIED COMMUNICATIONS MANAGEMENT' and displays a table of 'Elements'. The table has columns for Element Name, Element Type, Release, and Address. The element 'NRSM on sps3' is highlighted with a red box.

Element Name	Element Type	Release	Address
EM on nd2-carrier2	CS1000	6.0	47.248.100.138
EM on nd2-ss	CS1000	6.0	47.248.100.138
sip1-6.interop.com (member)	Linux Base	6.0	47.248.100.131
nd2-ss.interop.com (member)	Linux Base	6.0	47.248.100.253
nd2-carrier2.interop.com (primary)	Linux Base	6.0	47.248.100.251
sps3.interop.com (member)	Linux Base	6.0	47.248.100.150
47.248.100.137	Media Gateway Controller	6.0	47.248.100.137
NRSM on sps3	Network Routing Service	6.0	47.248.100.150

Figure 93 – Launch NRS Manager.

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

The screenshot shows the Nortel Network Routing Service Manager (NRS Manager) interface. The left sidebar contains a navigation menu with categories like System, Numbering Plans, and Tools. The main content area is titled 'NETWORK ROUTING SERVICE MANAGER' and displays the 'Domains' configuration page. The 'Service Domains (1)', 'L1 Domains (UDP) (1)', and 'L0 Domains (CDP) (1)' buttons are highlighted with red boxes. Below these buttons is a table with columns for Domain Name, Description, # of L1 Domains, and # of L0 Domains.

Domain Name	Description	# of L1 Domains	# of L0 Domains
1 interop.com	interop.com	1	1

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

Managing: Active database | 47.248.100.150
Standby database | Numbering Plans » Endpoints

Search for Endpoints

Enter an endpoint ID (use * for all) and click Search. You may narrow the search by specifying a particular domain.

Endpoint ID: *

Limit results to Domain: interop.com / udp / cdp

Gateway Endpoints (6)		User Endpoints (1)			
ID	Supported Protocols	SIP Mode	Call Signaling IP	Description	# of Routing Entries
1 car1_ss2	Dynamic SIP endpoint / NCS	Proxy Mode	Not available	car1_ss2	8
2 car2-ss3	Dynamic SIP endpoint / NCS	Proxy Mode	Not available	car2-ss3	0
3 car2_ss2	Dynamic SIP endpoint	Proxy Mode	Not available	car2_ss2	5
4 mp118_1	Dynamic SIP endpoint	Proxy Mode	Not available	mp118_1	1
5 mp118_mcs_usr	Dynamic SIP endpoint	Proxy Mode	Not available	mp118_mcs_usr	1

1 - 6 of 6 Gateway Endpoint(s) | Page 1 of 1

Figure 95 – NRS – Endpoint Creation.

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

Managing: Active database | 47.248.100.150
Standby database | Numbering Plans » Endpoints » Gateway Endpoint

Edit Gateway Endpoint (interop.com / udp / cdp)

End point name: car1_ss2 *

Description: car1_ss2

Trust Node:

Tandem gateway endpoint name: Not Applicable

Endpoint authentication enabled: Authentication off

Authentication password: _____

E.164 country code: 1

E.164 area code: 613

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**

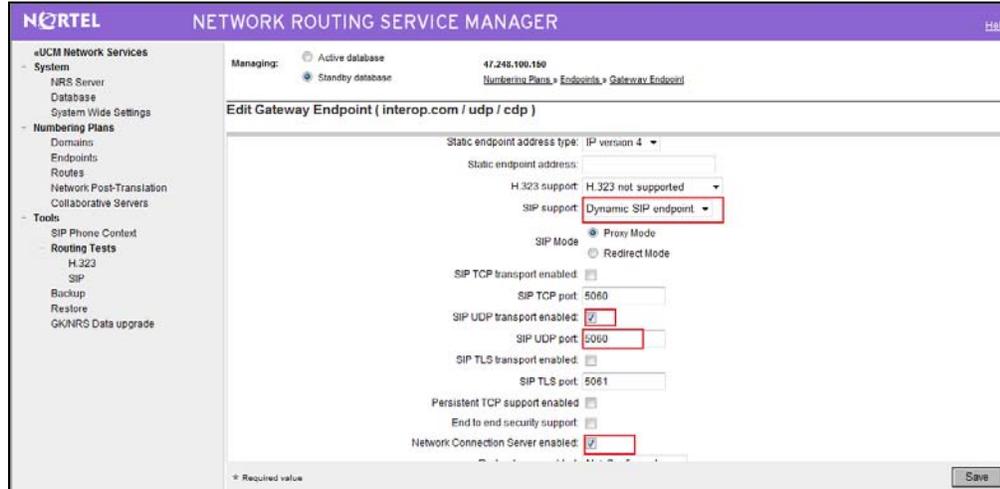


Figure 97 – Gateway Endpoint Configuration Details.

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.

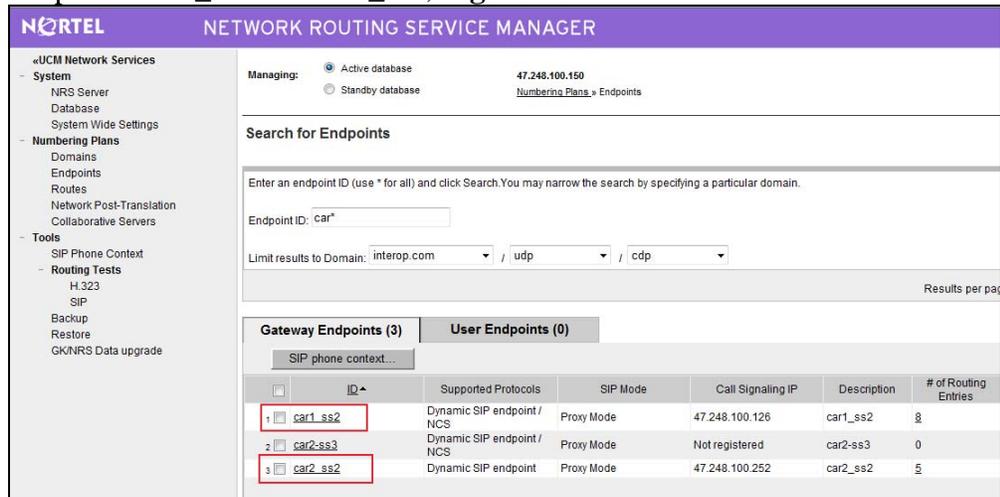


Figure 98 – Gateway Endpoint Details.

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.

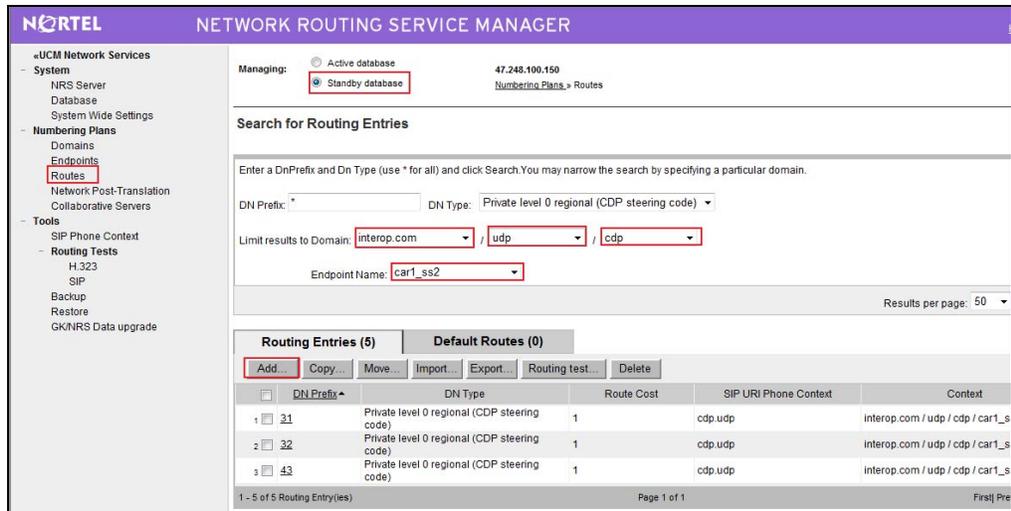


Figure 99 – Routes Configuration Page.

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).

DN Prefix	DN Type	Route Cost	SIP URI Phone Context	Context
1613	E.164 national	1	+1	interop.com / udp / cdp / car1_ss2
53	Private level 0 regional (CDP steering code)	1	cdp.udp	interop.com / udp / cdp / car1_ss2

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_B (car1_ss2).

DN Prefix	DN Type	Route Cost	SIP URI Phone Context	Context
1613	E.164 national	1	+1	interop.com / udp / cdp / car1_ss2
50	Private level 0 regional (CDP steering code)	1	cdp.udp	interop.com / udp / cdp / car2_ss2
53	Private level 0 regional (CDP steering code)	1	cdp.udp	interop.com / udp / cdp / car1_ss2
613	F.164 national	1	+1	interop.com / udp / cdp / node3_100

Figure 103 – Routes Entries for CS1000_B (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**.

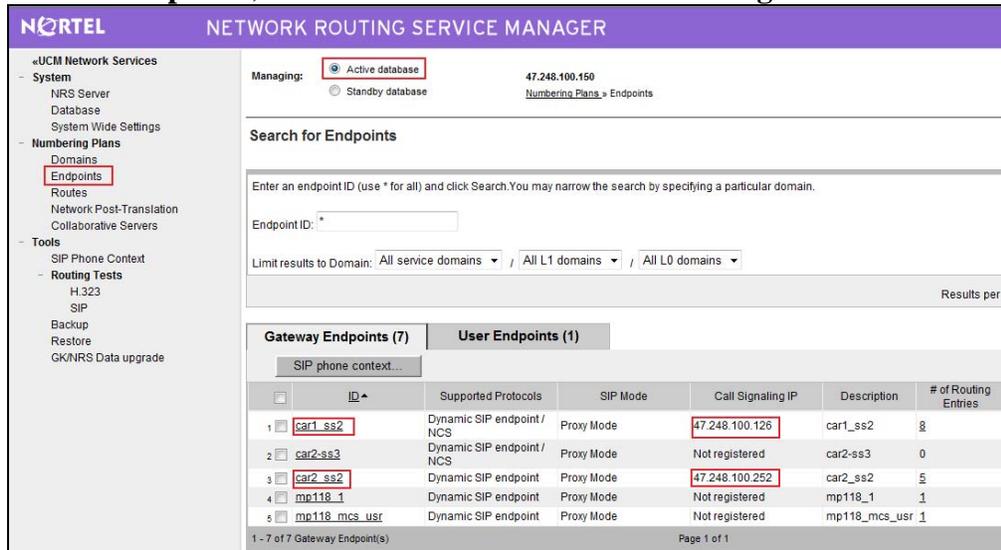


Figure 106 – Active Database.

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.

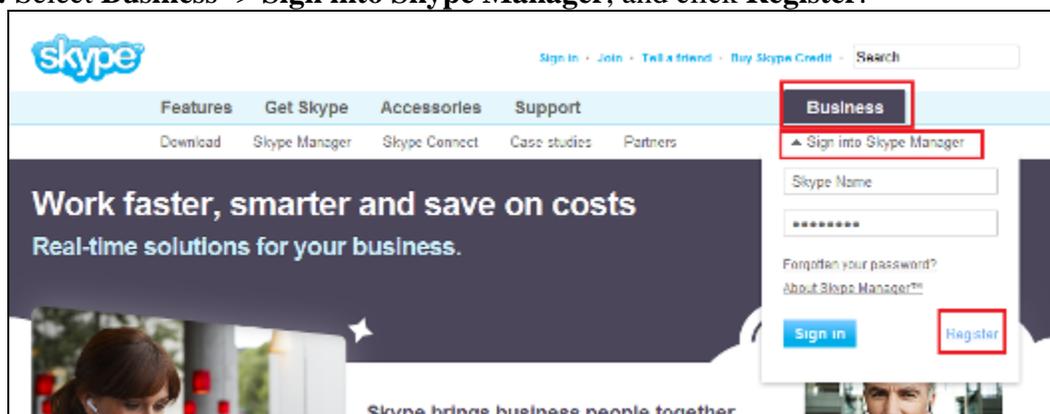


Figure 107 – Skype Registration

Step 2: Click on “Sign up now”

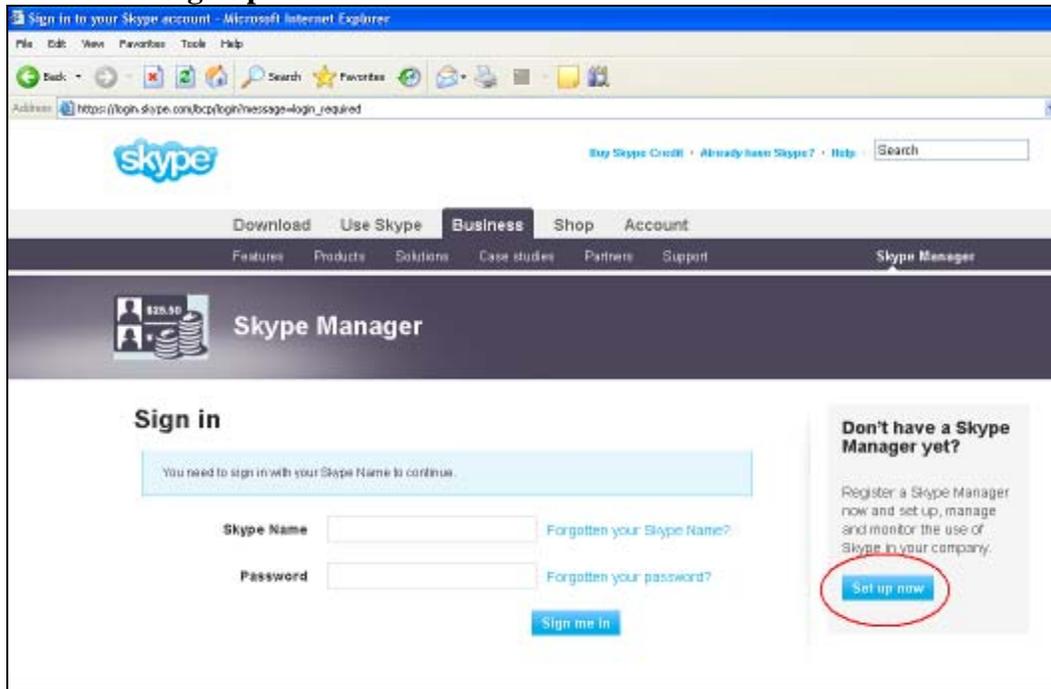


Figure 108 – Skype Registration (cont.)

Step 3: Select “No, I don’t have a Skype account”.

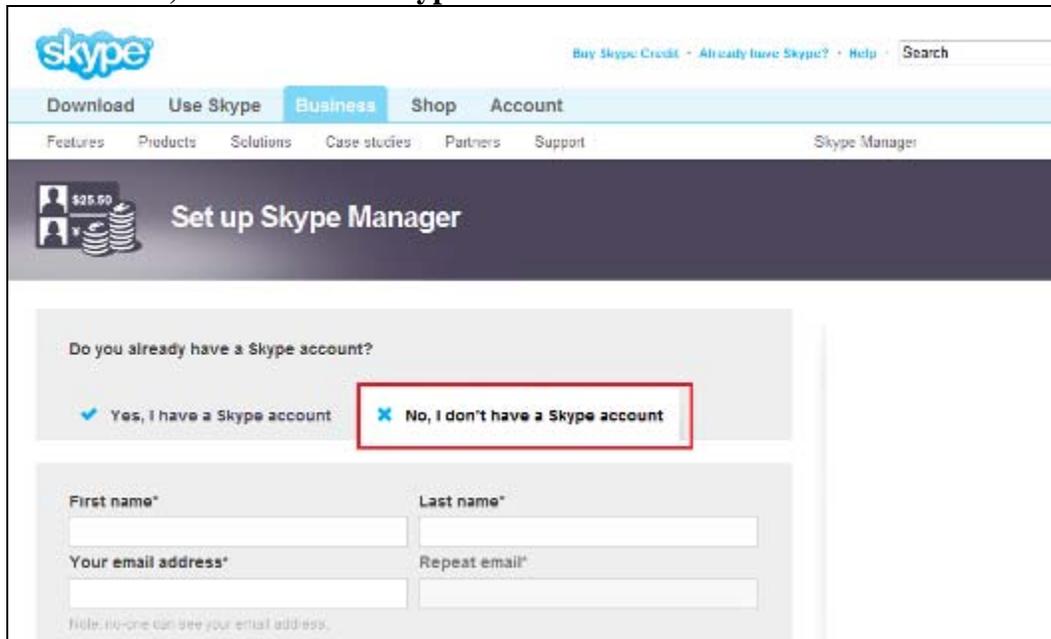


Figure 109 – Skype Registration (cont.)

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

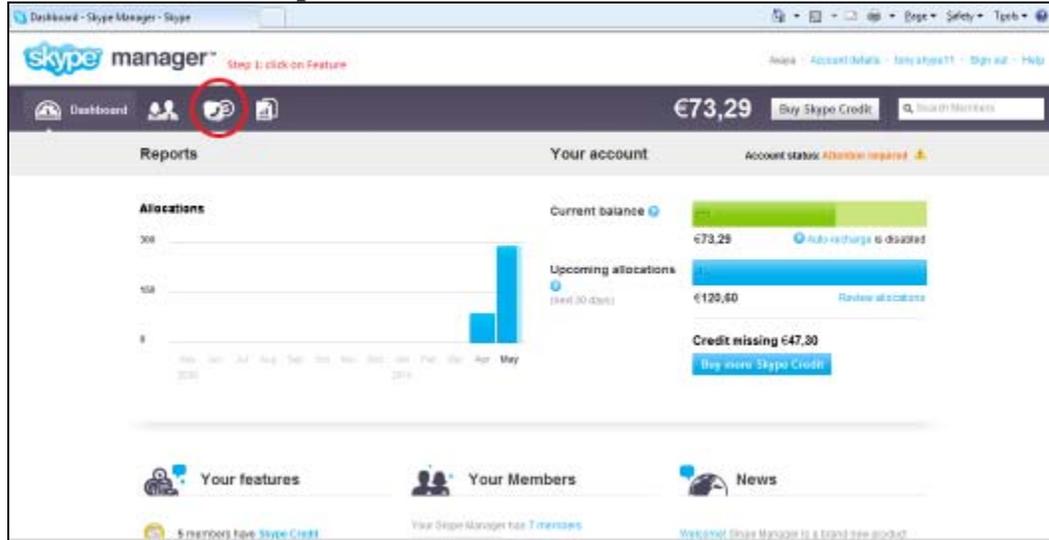


Figure 110 – Skype Registration (cont.)

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.

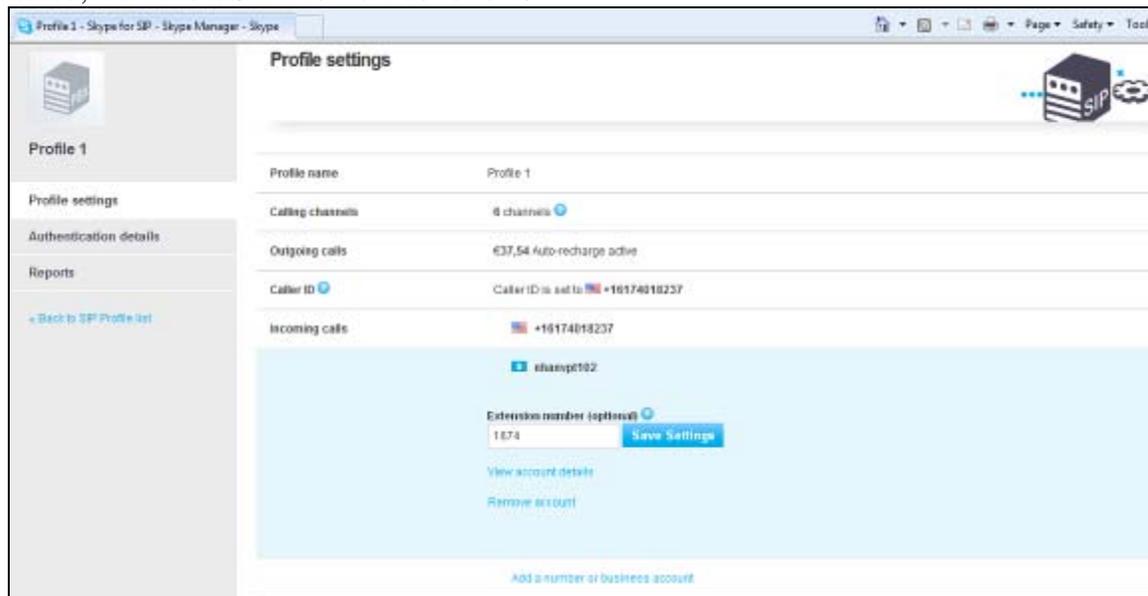


Figure 111 – Skype Profile Settings

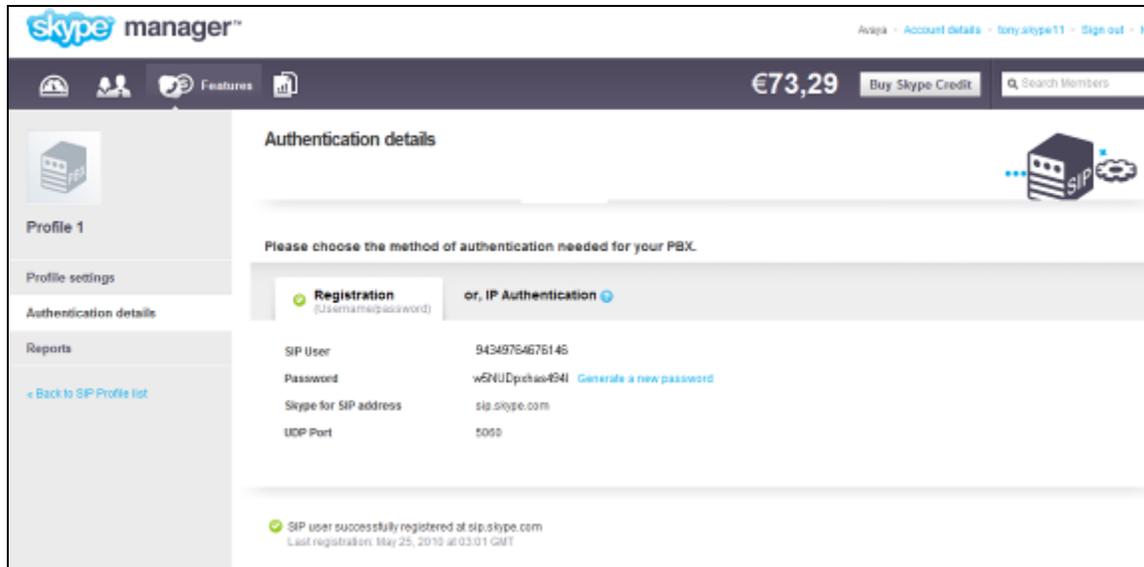


Figure 112 – Authentication Details

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.

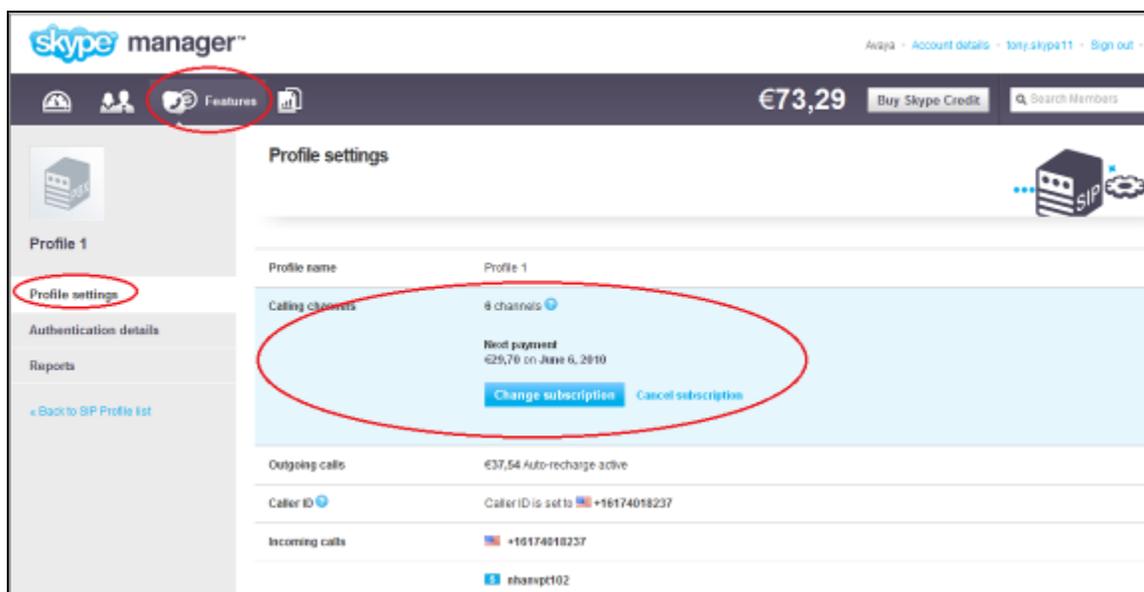


Figure 113 – Subscription in Profile Settings

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.

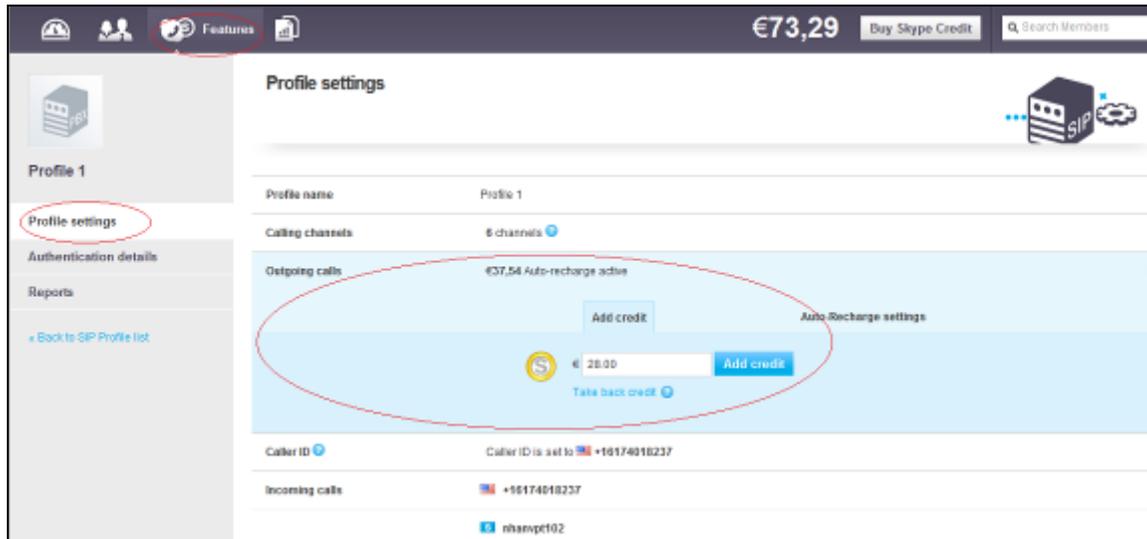


Figure 114 – Account Credit for Outbound Calling

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.

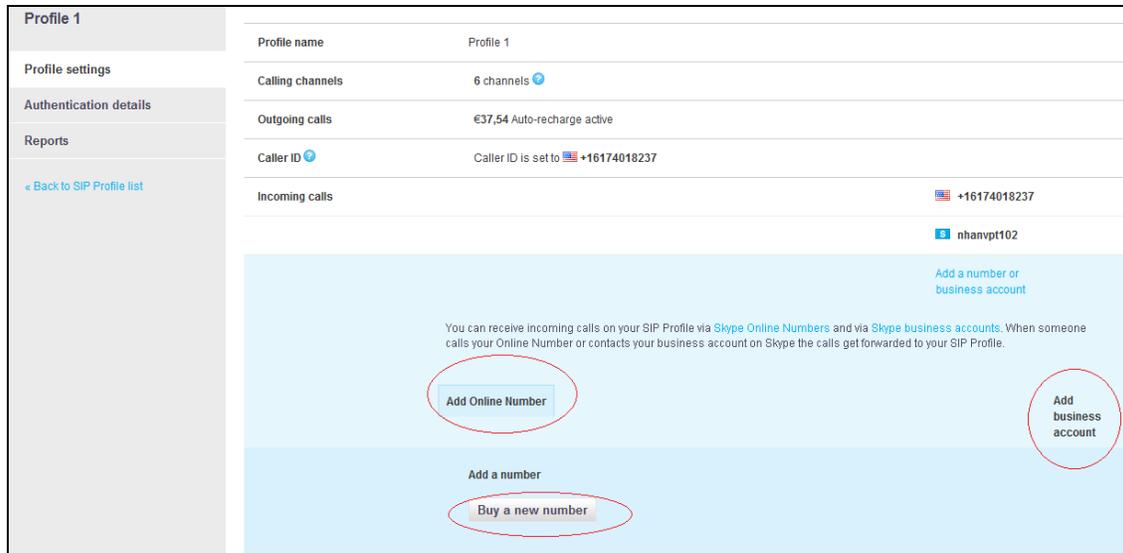


Figure 115 – SIP Profile for Inbound Calling

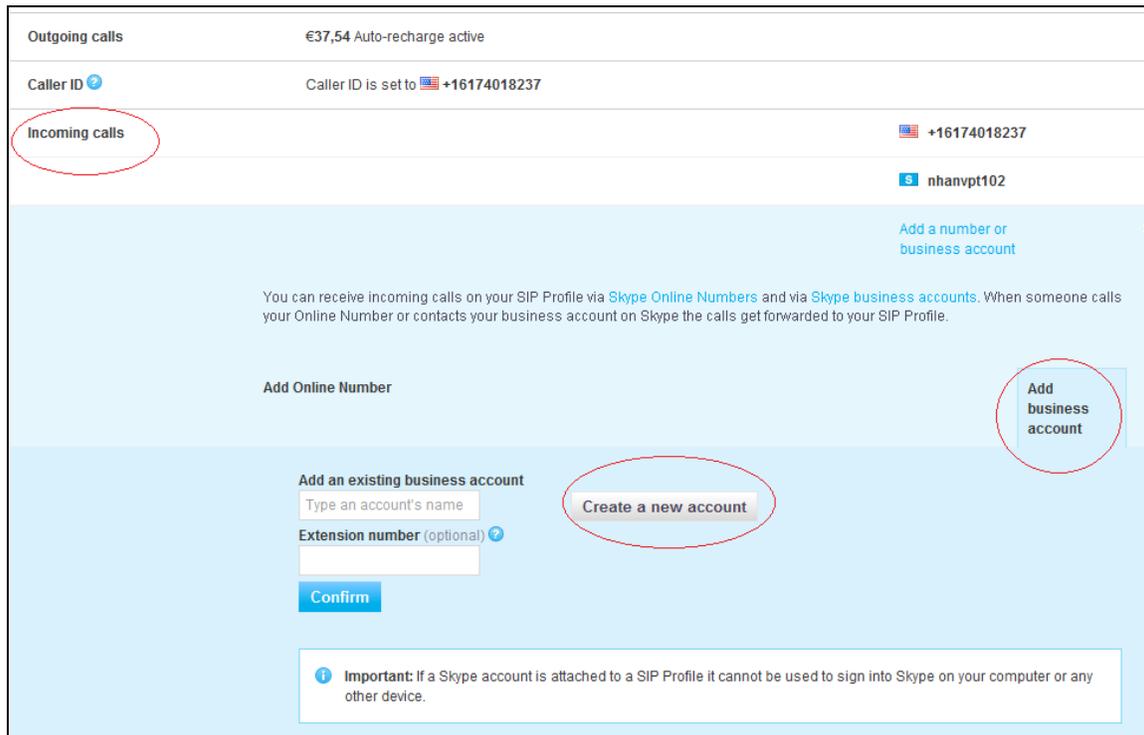


Figure 116 – SIP Profile for Inbound Calling (cont.)

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

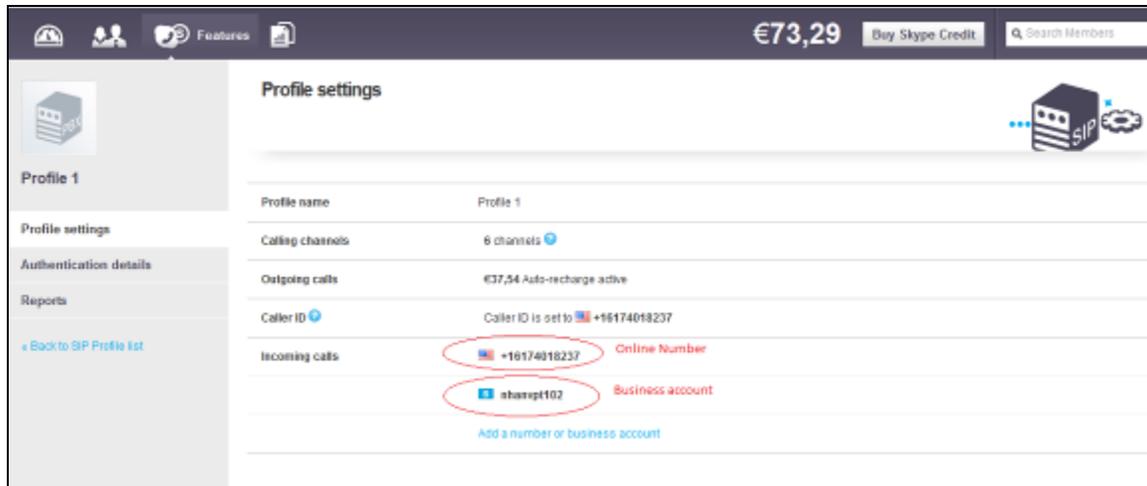


Figure 117 – SIP Profile for Inbound Calling (cont.)

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.

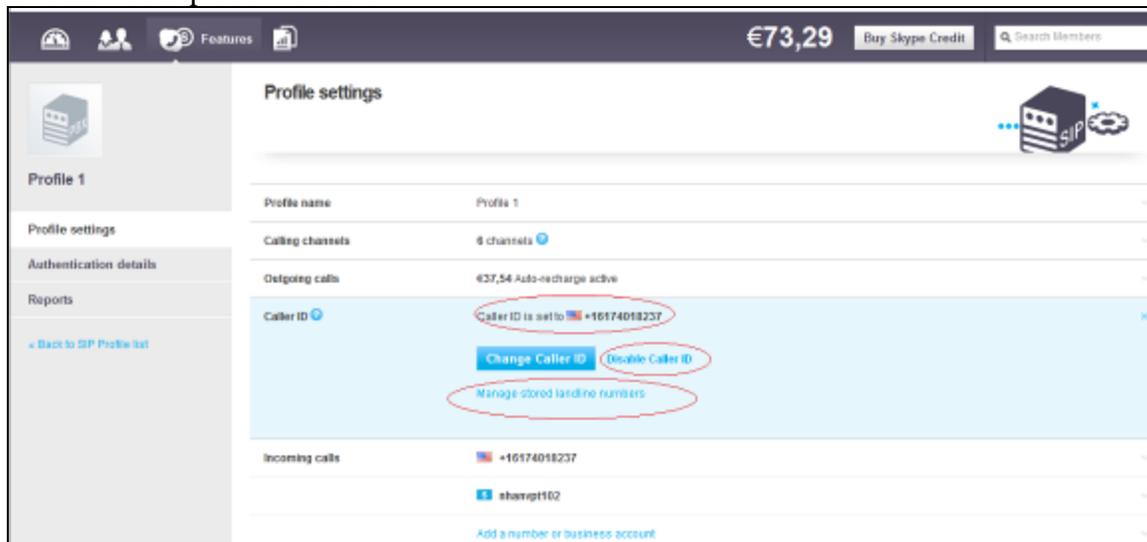


Figure 118 – SIP Profile for Outbound Calling

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,

Issue 01. 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 "Temporarily 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	C60

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-----Digital 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
  
```

Verify 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:IOF1	p28579_1	27/08/2010	p28579_1.cpm	NO
019	Q02033139	ISS1:IOF1	p28582_1	27/08/2010	p28582_1.cpm	NO
020	Q01782930-01	ISS1:IOF1	p24964_1	27/08/2010	p24964_1.cpm	NO
021	Q02018384	ISS1:IOF1	p28598_1	27/08/2010	p28598_1.cpm	NO
022	Q02033201	ISS1:IOF1	p28631_1	27/08/2010	p28631_1.cpm	YES
023	Q02089407	ISS1:IOF1	p29311_1	27/08/2010	p29311_1.cpm	NO
024	Q02038675	ISS1:IOF1	p28665_1	27/08/2010	p28665_1.cpm	YES
025	Q02020734-02	ISS1:IOF1	p28668_1	27/08/2010	p28668_1.cpm	NO
026	Q02038440	ISS1:IOF1	p28674_1	27/08/2010	p28674_1.cpm	NO
027	Q02035396	ISS1:IOF1	p28675_1	27/08/2010	p28675_1.cpm	NO
028	Q02038482	ISS1:IOF1	p28682_1	27/08/2010	p28682_1.cpm	NO
029	Q02039994	ISS1:IOF1	p28690_1	27/08/2010	p28690_1.cpm	NO
030	Q02024455-01	ISS1:IOF1	p28717_1	27/08/2010	p28717_1.cpm	NO
031	Q02031359	p28679	p28725_1	27/08/2010	p28725_1.cpm	YES
032	Q02083694	ISS1:IOF1	p29741_1	01/09/2010	p29741_1.cpm	NO
034	Q02108554	ISS1:IOF1	p29534_1	27/08/2010	p29534_1.cpm	NO
036	Q01974383-02	ISS1:IOF1	p27378_1	27/08/2010	p27378_1.cpm	NO
037	Q02092594	ISS1:IOF1	p27830_1	27/08/2010	p27830_1.cpm	NO
038	Q01999478-01	ISS1:IOF1	p27897_1	27/08/2010	p27897_1.cpm	NO
040	Q02007976-03	ISS1:IOF1	p28028_1	27/08/2010	p28028_1.cpm	NO
041	Q02007476	ISS1:IOF1	p28031_1	27/08/2010	p28031_1.cpm	NO
042	Q02011613-01	ISS1:IOF1	p28108_1	27/08/2010	p28108_1.cpm	NO
043	Q02017013-01	ISS1:IOF1	p28313_1	27/08/2010	p28313_1.cpm	NO
044	Q02097631	ISS1:IOF1	p28328_1	27/08/2010	p28328_1.cpm	NO
045	Q01987270-02	ISS1:IOF1	p28416_1	27/08/2010	p28416_1.cpm	NO
046	Q01938235-05	ISS2:IOF1	p28418_2	27/08/2010	p28418_2.cpm	NO
047	Q02032955-02	ISS1:IOF1	p28529_1	27/08/2010	p28529_1.cpm	NO
048	Q02019323-01	ISS1:IOF1	p28551_1	27/08/2010	p28551_1.cpm	NO
049	Q02100914	ISS1:IOF1	p28597_1	27/08/2010	p28597_1.cpm	NO
050	Q02032155	p28538	p28638_1	27/08/2010	p28638_1.cpm	YES
051	Q02040015	ISS1:IOF1	p28657_1	27/08/2010	p28657_1.cpm	NO
052	Q02031118	ISS1:IOF1	p28680_1	27/08/2010	p28680_1.cpm	NO
053	Q02029228-01	ISS1:IOF1	p28681_1	27/08/2010	p28681_1.cpm	NO
054	Q02043231	ISS1:IOF1	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	ISS1:1OF1	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-6.00.18.00-00.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-ftpkg-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_TR K	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=b0f360-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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