

Avaya Solution & Interoperability Test Lab

# Application Notes for Skype Connect Service with Avaya<sup>TM</sup> 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<sup>TM</sup> 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.

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# 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

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# 3. Equipment and Software Validated

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

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

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

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

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

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

[nortel@nodel-carrier ~]\$ pstat									
Product Release: 6.00.18.00									
In system patches: 6									
PATCH#	NAME	IN_SERVICE	DATE	SPECINS	TYI	PE RPM			
19	p28774_1	Yes	10/08/10	NO	FRU	U nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386			
20	p28797_1	Yes	10/08/10	NO	FRU	U nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386			
21	p29703_1	Yes	10/08/10	NO	FRU	U nortel-cs1000-shared-ssSubagent-6.00.18-00.i386			
22	p25946_1	Yes	10/08/10	NO	FRU	U nortel-cs1000-pi-control-1.00.00.00-00.noarch			
24	p27159_1	Yes	10/08/10	NO	FRU	U nortel-cs1000-pi-control-1.00.00.00-00.noarch			
26	p30291_1	Yes	10/08/10	NO	FRI	U nortel-cs1000-pi-control-1.00.00.00-00.noarch			
In Syst	em service	updates: 19							
PATCH#	IN_SERVICE	DATE	SPECINS	REMOVABL	Е	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			

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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	1			
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
[norte	l@nodel-carrie	er ~]\$			

# 4. Avaya Communication Server 1000 Configuration

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

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

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

# 4.1. Log in to CS1000 System

- 4.1.1. UCM and EM
- 4.1.2. Call Server Overlay

# 4.2. Administer A Node IP Telephony

- 4.2.1. Obtain Node IP address
- 4.2.2. Administer TPS
- 4.2.3. Administer Quality of Service (QoS)
- 4.2.4. Synchronize The New Configuration

# 4.3. Administer Voice Codec

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

#### 4.4. Zones and Bandwidth Management

- 4.4.1. Create a zone for IP phones (zone 10)
- 4.4.2. Create a zone for virtual SIP trunk (zone 255)

# 4.5. Administer SIP Trunk Gateway

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

# 4.6. Administer Dialing Plans

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

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- 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

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- 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.

			K
			NØRTEL
Use this page to access the server by IP address, You will need to log in again when switching to another server, even if R is in this same security domain.	User ID;	admin	3
Important: Only accounts which have been prevously created in the primary security server are allowed. Expired or react passwords that normally must be changed during topin will fail authentication in this mode (use the link to manual password change instead). Local OS-authenticated User IDs cannot be used.	Password	•••••	1
Go to central locin for Single Sign-On		Change Passwo	rd

Figure 3 – Log in Unified Communications Management.

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

NØRTEL	UNIFIED COMMUNICATION	S MANAGEMENT						
— Network Elements	Host Name: 47.248.100.245 Software Version: 02.00.0055.00(3266) User Name admin							
CS 1000 Services IPSec Patches SNMP Profiles Secure FTP Token	Elements New elements are registered into the security framework, or may be added as simple hyperlinks. Click an element name to launch its management service.							
Software Deployment			_					
— User Services Administrative Users	1 EM on node1-carrier	CS1000	6.0	47.248.100.147				
External Authentication Password — Security Roles Policies Certificates Active Sessions — Tools Logs	2 EM on node1-enter	CS1000	6.0	47.248.100.147				
	3 47.248.100.147	Call Server	6.0	47.248.100.147				
	4 🔲 sipl-6.interop.com (member)	Linux Base	6.0	47.248.100.119				
	5 node1-enter.interop.com (member)	Linux Base	6.0	47.248.100.124				
	6 node1-carrier.interop.com (primary)	Linux Base	6.0	47.248.100.245				
	7 🔲 47.248.100.148	Media Gateway Controller	6.0	47.248.100.148				

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#### Figure 4 – Unified Communications Management Page.

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

N@RTEL	CS 1000 ELEMENT MANAGER
- UCM Network Services - Home - Links	Managing: <u>47.248.100.147</u> Username: admin System Overview
Links     Virtual Terminals     Virtual Terminals     System     Alarms     Maintenance     Core Equipment     Peripheral Equipment     IP Network     Interfaces     Emergency Services     Geographic Redundancy     Software	System Overview
	IP Address: 47.248.100.147 Type: Nortel Communication Server 1000E CPPM Version: 4021 Release: 600 R +
<ul> <li>Customers</li> <li>Routes and Trunks</li> </ul>	Active Sessions
– 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	

Figure 5 – Element Manager System Overview Page.

#### **Call Server Overlay**

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

#### login as: **nortel**

#### Nortel Networks Linux Base 6.00

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

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

QT; Reviewed: SPOC 12/01/2010 Solution & Interoperability Test Lab Application Notes ©2010 Avaya Inc. All Rights Reserved. 11 of 126 NN10000-110CS1K [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

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.

NØRTEL	CS 10	00 ELEME	NT MANAGER				
UCM Network Services     Home     Links     Virtual Terminals     System     Alarms     Maintenance	Managing: 47.248.100.147 Username: admin System » IP Network » IP Telephony Nodes IP Telephony Nodes Click the Node D to view or edit its properties. Add Import Export Delete Print   Refre						
+ Core Equipment	Node ID	Components	Enabled Applications	ELAN IP	TLAN IP	Status	
- IP Network - Nodes: Servers, Media Cards	<u>1000</u>	1	PD, Presence Publisher, Gateway ( SIPGw	-	47.248.100.244	Synchronized	
- Maintenance and Reports	<u>1001</u>	1	SIP Line, LTPS, Gateway ( SIPGw	-	47.248.100.126	Synchronized	
- Media Gateways - Zones	1002	1	SIP Li	-	47.248.100.120	Synchronized	
- Host and Route Tables - Network Address Translation (N/ - QoS Thresholds - Personal Directories - Unicode Name Directory	Show: Vodes	Component Ser	vers 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.

NØRTEL	CS 1000 EL	EMENT M	IANAGER				
- UCM Network Services - Home - Links - Virtual Terminals	Managing: 47.248.100.147 Userr System » IP Network » Node Details (ID: 1000	name: admin IP Telephony Nodes - PD, Presenc	ce Publisher, Ga	ateway ( SIPC	Gw)		,
- Vinda Ferninas     - 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 (N,     - QoS Thresholds     - Personal Directories     - Unicode Name Directory     + Interfaces     - Engineered Values     + Emergency Services     + Geographic Redundancy     + Software	Node ID: 10 Call Server IP Address: 47 Telephony LAN (TLAN) Node IP Address: 47 Subnet Mask: 25 IP Telephony Voice Gateway (VGW Quality of Service (Qo LAN	2.248.100.147 2.248.100.244 5.255.255.240 Node Properties 1) and Codecs S)	(0-9999)	Embedded LA Gateway IP a Subne Applica • <u>SIP Line</u> • <u>Terminal Prc</u> • <u>Gateway (SI</u>	N (ELAN) address: 47.248 at Mask: 255.255 tions (click to edit bxy Server (TPS) PGw)	100.129 * 5.255.224 * t configuration)	E
	* Required Value. Associated Signaling S	Servers & Car	ds			Save	Cancel
- Customers	Select to add 🔻 Add	] [Remove]	Make Leader				Print   Refresh
<ul> <li>Routes and Trunks</li> <li>Routes and Trunks</li> <li>D-Channels</li> <li>Digital Trunk Interface</li> </ul>	Hostname     More 1-carrier	Type Signaling Server	Deployed Application LTPS, Gateway, PD	15 E 4	LAN IP 7.248.100.149	TLAN IP 47.248.100.245	Role Leader
<ul> <li>Dialing and Numbering Plans</li> <li>Electronic Switched Network</li> </ul>	available in the servers list.	art of any other in ter	opnony node and deploy	yee application(s) ti	for match the servic	agay accorde for this h	out are

Figure 7 – Node Details Page.

### 4.2.2. Administer TPS

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



#### Figure 8 – Node Details Page – TPS.

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

NØRTEL	CS 1000 ELEMENT MANAGER						
- UCM Network Services - Home - Links - Virtual Terminals	Managing: 47.248.100.147 Username: admin System » P Network » <u>P Telephony Nodes</u> Node ID: 1000 - UNIStim Line Terminal Proxy Server (LTPS) Configuration Details						
- System + Alarms - Maintenance + Core Equipment - Peripheral Equipment	UNIStim Line Terminal Proxy Server: 🗷 Enable proxy service on this node						
- IP Network - Nodes: Servers, Media Cards	IP Address: 0.0.0.0						
- Maintenance and Reports	Full file path: download/firmwar						
- Zones	Server Account/User ID:						
- Network Address Translation (N/	Password:						
- QoS Thresholds - Personal Directories - Unicode Name Directory	DTLS						
+ Interfaces - Engineered Values	DTLS Policy: Off -						
+ Emergency Services + Geographic Redundancy	Options: 🖂 Client Authentication						
- Customers	Periodic Re-keying						
<ul> <li>Routes and Trunks</li> <li>Routes and Trunks</li> <li>D-Channels</li> </ul>							
- Digital Trunk Interface     - Dialing and Numbering Plans     - Electronic Switched Network	* Required Value. Note: Changes made on this page will NOT be Save Cancel						
- <u>Nodes: Servers, Media Cards</u> - Maintenance and Reports     - Media Gateways     - Zones     - Host and Route Tables     - Network Address Translation (N.     - QoS Thresholds     - Personal Directories     - Unicode Name Directory     Interfaces     - Engineered Values     + Emergency Services     + Geographic Redundancy     + Software     - Customers     - Routes and Trunks     - D-Channels     - Digital Trunk Interface     - Dialing and Numbering Plans     - Electronic Switched Network	Address: 0.0.0      Full file path: download/firmwar Server Account/User ID:     Password:  DTLS  DTLS  DTLS Policy: Off  Options: Client Authentication Periodic Re-keying  * Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved. Cancel						

**Figure 9 – TPS Configuration Details.** 

#### 4.2.3. Administer Quality of Service (QoS)

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

NØRTEL	CS 1000 E	LEMENT M	IANAGER			
- UCM Network Services - Home - Links - Virtual Terminals	Managing: 47.248.100.147 Use System » IP Network Node Details (ID: 100	rname: admin » <u>IP Telephony Nodes</u> 0 - LTPS, PD, F	Presence Publisher	, Gateway ( SIP	Gw)	
- 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 (N- - OoS Thresholds	Node ID: Call Server IP Address: Telephony LAN (TLAN) Node IP Address: Subnet Mask: IP Telephor • Voice Gateway (VG	47.248.100.147 47.248.100.244 255.255.255.240 ny Node Properties W) and Codecs	(0-9999) E	mbedded LAN (ELAN) Sateway IP address: Subnet Mask: Applications (cli SIP Line	47.248.100.129 255.255.255.224 ck to edit configura	tion)
- Personal Directories     - Unicode Name Directory     + Interfaces     - Engineered Values     + Emergency Services     + Geographic Redundancy     + Software     - Customers	Quality of Service (C     LAN     * Required Value.  Associated Signaling Select to add      Add	Servers & Car	r <b>ds</b> Make Leader	Terminal Proxy Servi Gateway (SIPGw)	er (125)	Save Cancel  Print   Refresh
- Routes and Trunks	Hostname	Type	Deployed Applications	EL AN IP	TLANIP	Role
- D-Channels     - Digital Trunk Interface     Digital Trunk Interface	node1-carrier      Note: Only server(s) that are not	Signaling Server t part of any other IP te	LTPS, Gateway, PD lephony node and deployed a	47.248.10 application(s) that match	0.149 47.248.1 the service(s) selecte	00.245 Leader
Electronic Switched Network	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.

NØRTEL	CS 1000 ELEMENT MANAGER
- UCM Network Services - Home - Links - Virtual Terminals	Managing: 47.248.100.147 Username: admin System » P Network » <u>P Telephony Nodes</u> Node ID: 1000 - Quality of Service (QoS)
- System + Alarms	Diffserv Codepoint (DSCP)
+ Core Equipment - Peripheral Equipment	Control Packets: 40 (0-63)
- IP Network     - <u>Nodes: Servers, Media Cards</u> - Maintenance and Reports	Voice Packets: 46 (0-63)
Maintenance and Reports     Media Gateways     Zornes     - Host and Route Tables     - Host and Route Tables     Network Address Translation (N/         - QoS Thresholds     - Personal Directories     - Unicode Name Directory     Interfaces     Engineered Values     Emergency Services     Geographic Redundancy     Software	VLAN Tagging: 802.1Q Support 802.1Q Bits Value (802.1P): 6 (0-7)
- Customers - Routes and Trunks	
Routes and Trunks     D-Channels     Digital Trunk Interface     Dialing and Numbering Plans     Electronic Switched Network     Flexible Code Restriction	* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved. Cancel

# 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.

NØRTEL	CS 1000 EL	EMENT M	IANAGER			
- UCM Network Services - Home - Links - Virtual Terminals	Managing: 47.248.100.147 User System » IP Network » Node Details (ID: 1000	name: admin PTelephony Nodes - PD, Presence	ce Publisher, Gate	eway ( SIPGw)		
- 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 (N/ - QoS Thresholds - Personal Directories	Node ID: 1 Call Server IP Address: 4 Telephony LAN (TLAN) Node IP Address: 4 Subnet Mask: 2 IP Telephon Voice Gateway (VGV Quality of Service (Qd	000 * 7.248.100.147 * 7.248.100.244 * 55.255.255.240 * y Node Properties V) and Codecs 25)	(0-9999)	Embedded LAN (ELAN) Gateway IP address: Subnet Mask: Applications (clic SIP Line Terminal Proxy Serve	47.248.100.129 255.255.255.255.224 :k to edit configura er (TPS)	* * tion)
<ul> <li>Unicode Name Directory</li> <li>Interfaces</li> <li>Engineered Values</li> <li>Emergency Services</li> <li>Geographic Redundancy</li> </ul>	LAN     * Required Value.  Associated Signaling	Servers & Car	ds	Gateway (SIPGw)	[	Save Cancel
+ Software - Customers	Select to add 👻 Add	Remove	Make Leader			Print   Refresh
<ul> <li>Routes and Trunks</li> <li>Routes and Trunks</li> </ul>	Hostname +	Туре	Deployed Applications	ELAN IP	TLAN IP	Role
- D-Channels - Digital Trunk Interface	node1-carrier	Signaling Server	LTPS, Gateway, PD	47.248.100	0.149 47.248.1	00.245 Leader
<ul> <li>Dialing and Numbering Plans</li> <li>Electronic Switched Network</li> </ul>	Note: Only server(s) that are not available in the servers list .	part of any other IP te	lephony node and deployed	d application(s) that match t	he service(s) selecte	d for this node are

**Figure 12 – Synchronize the new Configuration – Save.** 

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

NØRTEL	CS 1000 ELEMENT MANAGER
– UCM Network Services – Home – Links – Virtual Terminals	Managing: 47.248.100.147 Username: admin System » IP Network » IP Telephony Nodes Node Saved
<ul> <li>System</li> <li>Alarms</li> <li>Maintenance</li> <li>Core Equipment</li> <li>Peripheral Equipment</li> </ul>	Node ID: 1000 has been saved on the call server. The new configuration must also be transferred to associated servers and media cards.
<ul> <li>IP Network</li> <li><u>Nodes: Servers, Media Cards</u></li> <li>Maintenance and Reports</li> </ul>	Transfer Now You will be given an option to select individual servers, or transfer to all.
- media Gateways - Zones - Host and Route Tables - Network Address Translation (N/ - QoS Thresholds	Show Nodes You may initiate a transfer manually at a later time.

Figure 13 – Synchronize the new Configuration – Transfer.

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

NØRTEL		CS 1000 ELEMENT MANAGER				
- UCM Network Services - Home - Links - Virtual Terminals - System + Alarms	Mana Syl Selec resta	ging: 47.248.100.147 Userna System » IP Network » IP nchronize Configural ct components to synchronize th rt* of applications on affected s	ame: admin <u>Prelephony Nodes</u> tion Files (Node ID leir configuration files with ca erver(s) when complete.	<1000>) Il server data. This process tr	ansfers server INI files to selected components, and requires a	
- Maintenance + Core Equipment		Start Sync Cancel	Restart Applications		Print   Refresh	
<ul> <li>Peripheral Equipment</li> <li>IP Network</li> </ul>		<u>Hostname</u>	Туре	Applications	Synchronization Status	
- Nodes: Servers, Media Cards		node1-carrier	Signaling Server	LTPS, Gateway, PD	Sync required	
<ul> <li>Maintenance and Reports</li> <li>Media Gateways</li> <li>Zones</li> <li>Host and Route Tables</li> <li>Network Address Translation (N-QoS Thresholds</li> </ul>	* Aj H323 serv	plication restart is only required Gateway settings, network cor ers.	for initial system configuration nectivity related parameters	n or if changes have been ma like ports and IP address, enai	de to general LAN configurations, SNTP settings, SIP and bing or disabling services, or adding or removing application	

**Figure 14 – Synchronize the new Configuration – Start Sync.** 

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

NØRTEL	CS 1000 EL	EMENT MAN	AGER	
UCM Network Services     Home     Links     - Virtual Terminals     System	Managing: 47.248.100.147 Userna System » IP Network » <u>II</u> Synchronize Configura Select components to synchronize th restart* of applications on affected s	ame: admin <u>P Telephony Nodes</u> <b>tion Files (Node IE</b> heir configuration files with o server(s) when complete.	) <1000>) all server data. This process tra	ansfers server INI files to selected components, and requires a
- Maintenance + Core Equipment - Peripheral Equipment	Start Sync Cancel	Restart Applications	Applications	Print   Refresh Synchronization Status
<ul> <li>Nodes: Servers, Media Cards</li> <li>Maintenance and Reports</li> <li>Media Gateways</li> <li>Zones</li> <li>Host and Route Tables</li> <li>Network Address Translation (N</li> <li>QoS Thresholds</li> </ul>	Application restart is only required H223 Gateway settings, network co servers.	Signaling Server d for initial system configurat innectivity related parameter	LTPS, Gateway, PD ion or if changes have been ma s like ports and IP address, enal	Synchronized de to general LAN configurations, SNTP settings, SIP and pling or disabling services, or adding or removing application

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.

NØRTEL	CS 1000 E	LEMENT M	IANAGER			
- UCM Network Services - Home	Managing: 47.248.100.147 Use System » IP Network	rname: admin » IP Telephony Nodes				
- Links - Virtual Terminals	Node Details (ID: 100	0 - PD, Presend	e Publisher, Gatev	vay ( SIPGw)		
<ul> <li>- System</li> <li>+ Alarms</li> <li>- Maintenance</li> </ul>	Node ID: [	1000	(0-9999)			<u>^</u>
+ Core Equipment - Peripheral Equipment	Call Server IP Address:	47.248.100.147				
- IP Network	Telephony LAN (TLAN)		E	mbedded LAN (ELAN)		
<ul> <li><u>Nodes: Servers, Media Cards</u></li> <li>Maintenance and Reports</li> </ul>	Node IP Address:	47.248.100.244	(	Gateway IP address:	47.248.100.129	· ·
– Media Gateways – Zones	Subnet Mask: 2	255.255.255.240		Subnet Mask:	255.255.255.224	*
- Host and Route Tables	IP Telephor	ny Node Properties	_	Applications (clic	ck to edit configur	ation)
- Network Address Translation (N/	<ul> <li>Voice Gateway (VG)</li> </ul>	W) and Codecs	•	SIP Line		
- Personal Directories	<ul> <li>Quality of Service (C</li> </ul>	<u>loS)</u>	- ·	Terminal Proxy Serve	er (TPS)	
- Unicode Name Directory	• LAN		•	Gateway (SIPGw)		•
+ Interfaces - Engineered Values	* Required Value.					Save Cancel
+ Emergency Services + Geographic Redundancy + Software	Associated Signaling	Servers & Car	ds			
- Customers	Select to add 🔹 Add	Remove]	Make Leader			Print   Refresh
- Routes and Trunks	Hostname +	Type	Deployed Applications	ELAN IP	TLAN I	P Role
- D-Channels - Digital Trunk Interface	node1-carrier	Signaling Server	LTPS, Gateway, PD	47.248.10	0.149 47.248.	.100.245 Leader
- Dialing and Numbering Plans - Electronic Switched Network	Note: Only server(s) that are not available in the servers list .	t part of any other IP te	ephony node and deployed a	pplication(s) that match t	the service(s) select	ed for this node are

Figure 16 – Node Details - Voice Gateway and Codec.

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



Figure 17 – Voice Gateway and Codec Configuration Details.

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

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

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

b) Select **IP Network** -> **Media Gateways** configuration from the left pane , click **MGC** from the right pane.

NØRTEL	CS	1000 ELEME	ENT MANAGER	
- UCM Network Services - Home	Managing: <u>47.248.</u> System :	100.147 Username: admi » IP Network » Media Gate	in ways	
- Links - Virtual Terminals - System + Alarms - Maintenance + Core Equipment	Media Gat	teways	Reboot Delete Virtual Terminal More Actions -	
Protection     Prover     Presonal Directories     Proverse     Prov	•	004.00	IP Address 47.248.100.148	Zone Type 010 MGC

Figure 18 – Media Gateways Configuration Page.

<u> </u>	<u> </u>		
NØRTEL	CS 1000 ELEMENT MANAGE	R	
- UCM Network Services	- VGW and IP phone codec profile		
- Home	Enable echo canceller		
- Links			
- Virtual Terminals	Echo canceller tail delay	64  (milliseconds)	
- System	Enable dynamic attenuation	<b>V</b>	
- Maintenance			1
+ Core Equipment	voice activity detection threshold		(0-4 DBM)
- Peripheral Equipment	Idle noise level	0	(0-1 DBM)
- IP Network - Nodes: Servers, Media Cards			(0-1050)
- Maintenance and Reports	DTMF tone detection		
- Media Gateways	Enable low latency mode		
- Zones	Demonstration of the second se		
<ul> <li>Host and Route Tables</li> <li>Network Address Translation (N.</li> </ul>	Remove DTMF delay (squeich DTMF from TDM to IP)		
- QoS Thresholds	Enable modem/fax pass through mode		
- Personal Directories	Enable V 21 FAX tone detection		
- Unicode Name Directory	Lindble V.2 ITAX tone detection		
- Engineered Values	Fax TCF method	2 -	
+ Emergency Services	FAX maximum rate	14400 T (bar)	
+ Geographic Redundancy		(bps)	
+ Software	FAX playout nominal delay	100	(0-300 milliseconds)
- Routes and Trunks	FAX no activity timeout	20	
- Routes and Trunks	PAX no activity uneout	20	( 10 - 32000 milliseconds )
- D-Channels	FAX packet size	30 -	
- Digital Trunk Interface	Codec G711	Calaat	
- Dialing and Numbering Plans	+codec G/TI	Select	
- Flexible Code Restriction	- Codec G729A	Select 🗹	
<ul> <li>Incoming Digit Translation</li> </ul>	Codec name	G729A	
- Phones	Voice pauload size	20 -	
- Templates	voice payloau size	20 • (ms/frame)	
- Properties	Voice playout (jitter buffer) nominal delay	40 🔻	
- Migration	Modifications may cause changes to dependent settings		
- Tools	Voice playout (jitter huffer) maximum delay	80 -	
+ Backup and Restore - Call Server Initialization	voice playout gitter barrer, maximum delay	199. N	
- Date and Time	Modifications may cause changes to dependent settings	_	
+ Logs and reports	VAD	<b>V</b>	
- Security	+ Codec 6723.1	Calant	
+ Passwords + Policies	+ COURC 0723.1	Select	
+ Login Options	+ Codec T38 FAX	Select V	
	+QoS		

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

Figure 19 – Media Gateways Configuration Details Page.

/ 0		
- Zones	+Codec G729A Select 🗸	
- Network Address Translation (N/	+ Codec G723.1 Select	
- Personal Directories	+ Codec T38 FAX Select 🗹	
- Unicode Name Directory	- QoS	
- Engineered Values	Enable Nortel Automatic QoS	
+ Emergency Services		
+ Geographic Redundancy + Software	Diffserv codepoint(DSCP) control packets 40 (0 - 63)	
- Customers	Diffserv codepoint(DSCP) voice packets 46 (0-63)	
- Routes and Trunks	- Call Server LAN	
- D-Channels	Embedded LAN (ELAN) configuration	
- Digital Trunk Interface	Geographic redundancy	
- Electronic Switched Network		
- Flexible Code Restriction	Primary call server IP address 47.248.100.147	
- Incoming Digit Translation	Primary call server hostname Primary_CS	
- Templates	Signaling port 15000	
- Reports	Broadcast nort 15001	
- Migration	(1024 - 65535 )	
- Tools	Telephony LAN (TLAN) configuration	
<ul> <li>Backup and Restore</li> <li>Call Server Initialization</li> </ul>	Signaling port 5000	
- Date and Time	Voice port 5200 (1024 - 65535)	
+ Logs and reports - Security	Routes	
+ Passwords	Add Remove	
+ Policies + Login Ontions		
- Login options	CHICK Add to add routes to the IPMG	
	Save Cancel VGW Channels	
	* Mandatory fields of current configuration	

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

Figure 20 – Media Gateways Configuration Details Page – Save.

# 4.4. Zones and Bandwidth Management

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

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

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

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

b) Select IP Network -> Zones configuration from the left pane, click Bandwidth Zones

NØRTEL	CS 1000 ELEMENT MANAGER
- 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	Managing: <u>47.248.100.147</u> Username: admin System » IP Network » Zones Zones Zones Zones are used to group related information for either bandwidth or dial plan numbering purposes. Bandwidth Zones 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.
- <u>Zones</u> - Host and Route Tables - Network Address Translation (N/ - QoS Thresholds	

#### Figure 21 – Zones Page.

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

NØRTEL	CS 1000 ELEMENT MANAGER					
- UCM Network Services - Home Linke	Managing: <u>47.248.100.147</u> Username: admin System » IP Network » <u>Zones</u> » Bandwidth Zones					
- Virtual Terminals	Bandwidth Zones					
- System + Alarms						
- Maintenance	Maintenance					
- Peripheral Equipment	- Maintenance Commands for Zones (LD 117)					
- IP Network	Configuration					
– Modes, Servers, Media Cards – Maintenance and Reports – Media Gateways	- Configuration Spreadsheet					
- Zones	Browse Import					
<ul> <li>Host and Route Tables</li> <li>Network Address Translation (N/ OoS Thrasholds</li> </ul>	Please Choose the Bandwidth Zones 1					
- Personal Directories	+ Bandwidth Zones 2					
- Unicode Name Directory	+ Bandwidth Zones 3					
- Engineered Values	- Bandwidth Zones 10					
+ Emergency Services	- Zone Basic Property and Bandwidth Management					
+ Geographic Redundancy + Software	- Adaptive Network Bandwidth Management and CAC - Alternate Routing for Calls between IP Stations					
- Customers	- Branch Office Dialing Plan and Access Codes					
- Routes and Trunks	<ul> <li>Branch Office Time Difference and Daylight Saving Time Property</li> </ul>					
<ul> <li>Routes and Trunks</li> <li>D-Channels</li> </ul>	+ Bandwidth Zones 255					

Figure 22 – Bandwidth Zones Page.

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

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

NØRTEL	CS 1000 ELEMENT MANAGER				
- UCM Network Services - Home - Links	Managing: <u>47.248.100.147</u> Username: admin System » IP Network » <u>Zones</u> » <u>Bandwidth Zones</u> » Bandwidth Zones 10 » Zone Basic Property and Bandwidth Management				
- Virtual Terminals - System + Alarms	Zone Basic Property and Bandwidth Management				
- Maintenance	Input Description	Input Value			
+ Core Equipment - Peripheral Equipment	Zone Number (ZONE):	10			
- Nodes: Servers, Media Cards	Intrazone Bandwidth (INTRA_BW):	1000000			
- Maintenance and Reports - Media Gateways	Intrazone Strategy (INTRA_STGY):	Best Quality (BQ) 🔻			
- Zones	Interzone Bandwidth (INTER_BW):	1000000			
<ul> <li>Host and Route Tables</li> <li>Network Address Translation (N/ - OoS Thresholds</li> </ul>	Interzone Strategy (INTER_STGY):	Best Quality (BQ)			
- Personal Directories	Resource Type (RES_TYPE):	Shared (SHARED) -			
<ul> <li>Unicode Name Directory</li> <li>Interfaces</li> </ul>	Zone Intent (ZBRN):	MO (MO) 🔻			
- Engineered Values	Description (ZDES):				
+ Emergency Services + Geographic Redundancy + Software - Customers	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**.

NØRTEL	CS 1000 ELEMENT MANAGER	
- UCM Network Services - Home Linke	Managing: <u>47.248.100.147</u> Username: admin System » IP Network » <u>Zones</u> » <u>Bandwidth Zones</u> » Bandwidth Zones 255 » Zone Basic Pr	operty and Bandwidth Management
- Virtual Terminals - System + Alarms	Zone Basic Property and Bandwidth Management	
- Maintenance	Input Description	Input Value
<ul> <li>Core Equipment</li> <li>Peripheral Equipment</li> <li>IP Network</li> <li>Nodes: Servers, Media Cards</li> <li>Maintenance and Reports</li> <li>Media Gateways</li> </ul>	Zone Number (ZONE): Intrazone Bandwidth (INTRA_BW): Intrazone Strategy (INTRA_STGY):	255 1000000 Best Quality (BQ)
- Zones - Host and Route Tables - Network Address Translation (N. - QoS Thresholds - Personal Directories - Unicode Name Directory + Interfaces - Engineered Values + Emergency Services + Geographic Redundancy + Software	Interzone Bandwidth (INTER_BW): Interzone Strategy (INTER_STGY): Resource Type (RES_TYPE): Zone Intent (ZBRN): Description (ZDES): Submit Refresh Delete Cancel	1000000 Best Quality (BQ) v Shared (SHARED) v VTRK (VTRK) v

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.

NØRTEL	CS 1000 ELEMENT MANAGER
- UCM Network Services - Home	Managing: <u>47.248.100.147</u> Username: admin <u>Customers</u> » Customer 00 » Edit
- LINKS - Virtual Terminals	Edit
- System + Alarms	
- Maintenance	Basic Configuration
- Peripheral Equipment	Application Module Link
+ IP Network	Call Detail Recording
+ Interfaces - Engineered Values	Call Party Name Display
+ Emergency Services	Call Redirection
+ Geographic Redundancy	Centralized Attendant Service
- Customers	Controlled Class of Service
- Routes and Trunks	Feature Options
- Routes and Trunks	Feature Packages
- Digital Trunk Interface	Flexible Feature Codes
- Dialing and Numbering Plans	Intercept Treatments
<ul> <li>Electronic Switched Network</li> <li>Elexible Code Restriction</li> </ul>	ISDN and ESN Networking

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).

NØRTEL	CS 1000 ELEMENT MANAGER		
- UCM Network Services - Home - Links - Virtual Terminals - System + Alarms - Maintenance + Core Equipment - Peripheral Equipment + IP Network + Interfaces - Engineered Values + Emergency Services + Gographic Redundancy + Software	Digital Private Network Signaling System 1     Flexible Tones and Cadences     Multifrequency Compelled Signaling     International Supplementary Features     Enhanced Night Service     Integrated Services Digital Network     + Dial Access Prefix on CLID table entry option     Integrated Services Digital Network     - Virtual Private Network Identifier     - Private Network Identifier	Package: 123 Package: 125 Package: 128 Package: 131 Package: 133 Package: 145 1 1	(1 - 16383)
- Customers - Routes and Trunks - Poutes and Trunks - D-Channels - Digital Trunk Interface - Dialing and Numbering Plans - Electronic Switched Network - Elexible Code Restriction	- Node DN: - Multi-location Business Group: - Business Sub Group Consult-only; - Prefix 1:	0 65535	(0 - 65535) (0 - 65535)

Figure 26 – Customer – ISDN Configuration Page.

# 4.5.2. Administer SIP Trunk Gateway to Skype

QT; Reviewed: SPOC 12/01/2010

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)

NØRTEL	CS 1000 EL	EMENT M	ANAGER				
- UCM Network Services - Home - Links - Virtual Terminals	Managing: 47.248.100.147 User System » IP Network » Node Details (ID: 1000	name: admin <u>IP Telephony Nodes</u> - PD, Presenc	e Publisher, Ga	teway ( SIPC	Gw)		
- Virtual Terminals - System + Alarms - Maintenance + Core Equipment - Peripheral Equipment - IP Network - Nodes: Servers, Media Cards - Maintenance and Reports - Media Cateways - Zones - Network Address Translation (N QoS Thresholds - Personal Directories	Call Server IP Address: 4 Telephony LAN (TLAII) Node IP Address: 4 Subnet Mask: 2 IP Telephony Quality of Service (Qc LAN SNITP Numbering Zones	7.248.100.147 * 7.248.100.244 * 55.255.255.240 * y Node Properties y) and Codecs (S)	Ε	Embedded LA Gateway IP a Subne Applica SIP Line Terminal Pro Gateway (SI Personal Dir Presence Pi	IN (ELAN) address: 47.248. et Mask: 255.255 tions (click to edit oxy Server (TPS) IPGw) rectories (PD) ublisher	100.129 • 5.255.224 • configuration)	E
+ Interfaces - Engineered Values + Emergency Services + Geographic Redundancy + Software	*Required Value. Associated Signaling	Servers & Car	ds			Save	Cancel
- Customers	Select to add 🔻 Add	Remove	Make Leader			1	Print   Refresh
<ul> <li>Routes and Trunks</li> <li>Routes and Trunks</li> </ul>	☐ Hostname ▲	Туре	Deployed Application	<u>is</u> <u>E</u>	LAN IP	TLAN IP	Role
- D-Channels - Digital Trunk Interface	node1-carrier	Signaling Server	LTPS, Gateway, PD	4	7.248.100.149	47.248.100.245	Leader
- Dialing and Numbering Plans - Electronic Switched Network	Note: Only server(s) that are not p available in the servers list .	part of any other IP tel	ephony node and deploy	ed application(s) the	hat match the servic	e(s) selected for this n	ode are

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.

- Local SIP Port: 5060

- SIP User: 99051000106920
  - Password: xxxxxxxxxxxxxxx
  - Primary Skype for SIP IP: 204.9.161.164
- ---> Gateway password
  - ---> Proxy Primary TLAN IP Address
     ---> Proxy Secondary TLAN IP Address

---> Gateway Endpoint Name

- Secondary Skype for SIP IP: 63.209.144.201

NØRTEL	CS 1000 ELEMENT MANAGER
- UCM Network Services - Home - Links - Virtual Terminals Suctor	Managing: 47.248.100.147 Username: admin System » P Network » <u>P Telephony Nodes</u> Node ID: 1000 - Virtual Trunk Gateway Configuration Details
+ Alarms	General   SIP Gateway Settings   SIP Gateway Services
- Maintenance + Core Equipment - Peripheral Equipment - IP Network	Vtrk Gateway Application: 🖉 Enable gateway service on this Node
- <u>Nodes: Servers, Media Cards</u> - Maintenance and Reports	Vtrk Gateway Application: SIP Gateway (SIPGw) - Monitor IP Addresses (listed below)
<ul> <li>Media Gateways</li> <li>Zones</li> <li>Host and Route Tables</li> <li>Network Address Translation (N/ 0.05 Thresholds</li> </ul>	SIP Domain name: sip.skype.com
	Local SIP Port: 5060 *(1 - 65535) Monitor IP:
– Personal Directories – Unicode Name Directory	Gateway endpoint name: 99051000106920
+ Interfaces - Engineered Values	Gateway password:
+ Emergency Services + Geographic Redundancy + Software	Enable failsafe NRS:
- Customers	
<ul> <li>Routes and Trunks</li> <li>Routes and Trunks</li> </ul>	SIP Gateway Settings
- D-Channels - Digital Trunk Interface	TLS Security: Security Disabled -
- Dialing and Numbering Plans - Electronic Switched Network	* Required Value. Note: Changes made on this page will NOT be Save Cancel

Figure 28 – Virtual Trunk Gateway Configuration Details Page.

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

Primary TLAN IP Address:provided when user creates a SIP profile on Skype.Secondary TLAN IP Address:provided when user creates a SIP profile on Skype.Port:5060

Transport Protocol: UDP

<b>Options:</b>	Check Support	registration and	Primarv	<b>CDS Proxv</b>
opnono	check Support	i chisti atton and	· • • • • • • • • • • • • •	

NØRTEL	CS 1000 ELEMENT MANAGER	
- UCM Network Services	Managing: 47.248.100.147 Username: admin Svstem » IP Network » IP Telephony Nodes	
- Home - Links - Virtual Terminals	Node ID: 1000 - Virtual Trunk Gateway Configuration Details	
- System	General   SIP Gateway Settings   SIP Gateway Services	
+ Alarms	SIP Gateway Settions	
- Maintenance + Core Equipment - Peripheral Equipment	TLS Security: Security Disabled	
- IP Network	Port: 5061 (1 - 65535)	
- <u>Nodes: Servers, Media Cards</u> - Maintenance and Reports	Number of Byte Re-negotiation: 0 +	=
- Media Gateways	Options: Client Authentication	
<ul> <li>Host and Route Tables</li> <li>Network Address Translation (N/</li> </ul>	X509 certificate authority	
- QoS Thresholds	Proxy Or Redirect Server:	
<ul> <li>Personal Directories</li> <li>Unicode Name Directory</li> </ul>	Primary TLAN IP Address: 204.9.161.164 Secondary TLAN IP Address: 63.209.144.201	
- Engineered Values	Port: 5060 (1 - 65535) Port: 5060 (1 - 65535)	2
+ Emergency Services + Geographic Redundancy	Transport protocol: UDP - Transport protocol: UDP -	
+ Software	Ontions: Support registration	3.1
- Customers	Options. Support registration	
- Routes and Trunks	Primary CDS Proxy	
- D-Channels - Digital Trunk Interface	CLID Presentation:	-
- Dialing and Numbering Plans - Electronic Switched Network	* Required Value. Note: Changes made on this page will NOT be Save Save	Cancel

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### 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.

NØRTEL	CS 1000 ELEMENT MANAGER					
- UCM Network Services	Managing: 47.248.100.147 Username: admin System » IP Network » I <u>P Telephony Nodes</u>					
- Links - Virtual Terminals	Node ID: 1000 - Virtual Trunk Gateway Configuration Details					
- System	General   SIP Gateway Settings   SIP Gateway Services					
+ Alarms - Maintenance	Number Translation: Strip: Prefix: CLID Display Format:					
+ Core Equipment	Subscriber (SN): 0 <ccc><area code=""/><sn></sn></ccc>					
<ul> <li>Peripheral Equipment</li> <li>IP Network</li> </ul>	National (NN): 0 <ccc><nn></nn></ccc>					
- Nodes: Servers, Media Cards	International: 0					
Maintenance and Reports     Media Gateways     Zones     Host and Route Tables     Network Address Translation (Ni     QoS Thresholds     Personal Directories     Unicode Name Directory     Interfaces     Engineered Values     Emergency Services     Geographic Redundancy     Software	SIP URI Map: Public E.164 Domain Names Private Domain Names National: Subscriber: Unknown:					
- Customers	SIP Gateway Services					
- Routes and Trunks - D-Channels - Digital Trunk Interface	SIP Converged Desktop: Enable CD service Service DN: Used for making VTRK call from agent.					
- Dialing and Numbering Plans - Electronic Switched Network	* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved. Save	Cancel				

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.

🕰 🔐 🥑 Features		€101,29	Buy Skype Credit	<b>Q</b> Search Members
BIRDEX	Authentication detail	ls		B SIP
Gubineiz Freite F	Please choose the method	of authentication ne	eded for your PBX.	
Profile settings	Ba window finan			
Authentication details	(Username/password)	or, IP Authentica	tion 😢	
Reports	SIP User			
« Back to SIP Profile list	Password Skype for SIP address UDP Port	sip.skype.com 5060	Generate a new password	
(	SIP user successfully registere Last registration: May 19, 2010	ed at sip.skype.com at 21:35 GMT		

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.** 

NØRTEL	CS 1000	ELEMENT MA	NAGER		
- UCM Network Services - Home	Managing: <u>47.248.100.147</u> Routes and Trunks	Jsername: admin » D-Channels			
- Links - Virtual Terminals - System + Alarms - Maintenance	D-Channels				
+ Core Equipment - Peripheral Equipment - IP Network - Nodes: Servers, Media Cards - Maintenance and Reports - Media Gateways - Zones	D-Channel Diagnostics (LD 96) Network and Peripheral Equipment (LD 32, Virtual D-Channels) MSDL Diagnostics (LD 96) TMDI Diagnostics (LD 96) D-Channel Expansion Diagnostics (LD 48)				
- Host and Route Tables - Network Address Translation (N/ - QoS Thresholds - Personal Directories - Unicode Name Directory + Interfaces	Configuration Choose a D-Channel Number: 0  and type: DCH  to Add				
<ul> <li>Engineered Values</li> <li>Emergency Services</li> </ul>	- Channel: 11	Type: DCH	Card Type: DCIP	Description: SIPL	Edit
+ Geographic Redundancy + Software	- Channel: 100	Type: DCH	Card Type: DCIP	Description: VoIP	Edit
- Customers     - Routes and Trunks     - Bottes and Trunks     - D-Channels     - Digital Trunk Interface	- Channel: 101	Type: DCH	Card Type: DCIP	Description: Enterprise	Edit

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



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#### 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.** 

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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.

NØRTEL	CS 1000 ELE	MENT MANA	
- UCM Network Services	Managing: <u>47.248.100.147</u> Username: admin System » Core Equipment » Superloops Superloops Add Delete		
- Core Equipment     - Loops     - <u>Superloops</u> - MSDL/MISP Cards     - Conference/TDS/Multifrequen     - Tone Senders and Detectors     - Peripheral Equipment     - IP Network     - Nodes: Servers, Media Cards     - Maintenance and Reports	Superloop Number ▲         1 ○ 4         2 ○ 96         3 ○ 100         4 ○ 104         5 ○ 124	Superloop Type IPMG Virtual Virtual Virtual Virtual Virtual	

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.

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



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- D-Channels	- Display of ac	cess prefix on CLID (DAPC) 📃
<ul> <li>Digital Trunk Interface</li> <li>Dialing and Numbering Plans</li> </ul>	- Basic Route Options	
- Electronic Switched Network	Attendant announcement (ATAN)	No Attendant Announcement. (NO)
<ul> <li>Flexible Code Restriction</li> <li>Incoming Digit Translation</li> </ul>	Billing number required (BILN)	
- Phones	Call detail recording (CDR)	
- Reports	North American toll scheme (NATL)	
- Properties - Migration	Controls or timers (CNTL)	
- Tools	Conventional (Tie trunk only) (CNVT)	
+ Backup and Restore - Call Server Initialization	Incoming DID digit conversion on this route (IDC)	
<ul> <li>Date and Time</li> <li>Logs and reports</li> </ul>	- Day IDC tree number (DCNO)	1 Range: 0 - 254
- Security	- Night IDC tree number (NDNO)	1 Range: 0 - 254
+ Passwords + Policies	- Display external dialed digits (DEXT)	

#### **Figure 38 – Route Configuration Details Page.**

Figure 39 – Route Configuration Details Page.

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

# 4.5.6. Administer Virtual Trunks

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

b) Continue Section 4.5.5, after click **Submit**, the **Routes and Trunks** screen is displayed and updated with the newly added route. Click the **Add trunk** button next to the newly added route.

NØRTEL	CS 1000	ELEMENT MA	ANAGER	
- UCM Network Services	Managing: <u>47.248.100.147</u> Us Routes and Trunks »	sername: admin Routes and Trunks		
Links - Virtual Terminals System + Alarms	Routes and Trur	nks		
- Maintenance - Core Equipment	- Customer: 0	Total routes: 4	Total trunks: 96	Add route
- Loops - Superloops	+ Route: 11	Type: TIE	Description: SIPL	Edit Add trunk
- MSDL/MISP Cards - Conference/TDS/Multifrequen	+ Route: 100	Type: TIE	Description: CARRIER	Edit Add trunk
- Tone Senders and Detectors - Peripheral Equipment	+ Route: 101	Type: TIE	Description: ENTERPRISE	Edit Add trunk
<ul> <li>IP Network</li> <li>Nodes: Servers, Media Cards</li> <li>Maintenance and Reports</li> </ul>	- Route: 105	Type: DID	Description: 911	Edit 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)

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- 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

NØRTEL	CS 1000 ELEMENT MANAGER			
- UCM Network Services	Managing: <u>47.248.109.147</u> Username: admin Routes and Trunks » <u>Routes and Trunks</u> » Customer 0, Route 100, New Trunk Configuration			
- Virtual Terminals	Customer 0, Route 100, New Trunk Configuration			
- System				
+ Alarms				
- Core Equipment	- Basic Configuration			
-Loops	Input Description		Input Value	
- Superloops - MSDL/MISP Cards	Multiple trunk input number (MTINPUT)	32 👻		
- Conference/TDS/Multifrequen	Trunk data block (TYPE)	IP Trunk (IPTI)	•	
<ul> <li>For a sender send betectors</li> <li>Peripheral Equipment</li> </ul>	Terminal Number (TN)	100 00 01 00	•	
- IP Network	Designator field for trunk (DES)	Carrier		
- Maintenance and Reports	Extended Trunk (XTRK)	VTRK		
- Media Gateways	Poute number Nember number (PTHP)	100.1	-	
- Host and Route Tables	Koute number (Krimb)	100 1		
- Network Address Translation	Level 3 Signaling (SIGL)			•
- QoS Thresholds	Card Density (CDEN)		10	
- Unicode Name Directory	Start arrangement Incoming (STRI)	Immediate (IMM)		•
+ Interfaces - Engineered Values	Start arrangement Outgoing (STRO)	Immediate (IMM)		
+ Emergency Services	Trunk Group Access Restriction (TGAR)	0		
+ Geographic Redundancy + Software	Channel ID for this trunk (CHID)	1		
- Customers				
- Routes and Trunks	Increase or decrease the member numbers (INC)	Increase channel	and member number (YES)	•
- Routes and Trunks	Class of Service (CLS)	Edit		
- D-Channels - Digital Trunk Interface	- Advanced Trunk Configurations			

Figure 41 – new Trunk Configuration Details Page.

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

of Server and then click Save (not shown).



Figure 42 – Class of Service Configuration Details Page.

# 4.6. Administer Dialing Plans

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

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

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

NØRTEL	CS 1000 ELEMENT MANAG
- UCM Network Services	Managing: <u>47.248.100.147</u> Username: admin Dialing and Numbering Plans » Electronic Switched Network (
- Virtual Terminals - System + Alarms	Electronic Switched Network (ESN)
- Mantenance     - Core Equipment     - Loops     - Superloops     - MSDL/MISP Cards     - Conference/TDS/Multifrequen     - Tone Senders and Detectors     - Peripheral Equipment     - IP Network     - Nodes: Servers, Media Cards     - Maintenance and Reports     - Media Gateways	- Customer 00     - Network Control & Services         - Network Control Parameters (NCTL)         - ESN Access Codes and Parameters (ESN)         - Digit Manipulation Block (DGT)         - Route List Block (RLB)         - Incoming Trunk Group Exclusion (ITGE)         - Network Attendant Services (NAS)         - Coordinated Dialing Plan (CDP)         - Local Steering Code (LSC)         - Distant Steering Code (DSC)
<ul> <li>Zones</li> <li>Host and Route Tables</li> <li>Network Address Translation</li> <li>QoS Thresholds</li> <li>Personal Directories</li> <li>Unicode Name Directory</li> <li>Interfaces</li> <li>Engineered Values</li> <li>Emergency Services</li> <li>Geographic Redundancy</li> <li>Software</li> </ul>	<ul> <li>Trunk Steering Code (TSC)</li> <li>Numbering Plan (NET)         <ul> <li>Access Code 1</li> <li>Home Area Code (HNPA)</li> <li>Home Location Code (HLOC)</li> <li>Location Code (LOC)</li> <li>Numbering Plan Area Code (NPA)</li> <li>Exchange (Central Office) Code (NXX)</li> <li>Special Number (SPN)</li> <li>Network Speed Call Access Code (NSCL)</li> <li>Eree Calling Area Screening (ECAS)</li> </ul> </li> </ul>
- 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	Free Calify Alea Screening (FCAS)     Free Special Number Screening (FSNS)     Access Code 2     Home Area Code (HNPA)     Home Location Code (HLOC)     Location Code (LOC)     Numbering Plan Area Code (NPA)     Exchange (Central Office) Code (NXX)     Special Number (SPN)     Network Speed Call Access Code (NSCL)

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** 

NØRTEL	CS 1000 ELEMENT MANAGER
- UCM Network Services - Home	Managing: <u>47.248.100.147</u> Username: admin Dialing and Numbering Plans » <u>Electronic Switched Network (ESN)</u> » Customer 00 »
- Virtual Terminals	Digit Manipulation Block List
+ Alarms - Maintenance + Core Equipment	Please Choose the Digit Manipulation Block Index 7 👻 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.**
NØRTEL	CS 1000 ELEMENT MANAGER
- UCM Network Services     - Home     - Links     - Virtual Terminals	Managing: <u>47.248.100.147</u> Username: admin Dialing and Numbering Plans » <u>Electronic Switched Network (ESN)</u> » Customer 00 » Network Control & Services » <u>Digit M</u>
- System + Alarms - Maintenance	Input Description
- Core Equipment - Loops - Superloops - MSDL/MISP Cards	Digit Manipulation Index numbers (DMI): 7 Number of leading digits to be Deleted (DEL) 7 (0-19)
<ul> <li>Conference/TDS/Multifrequen</li> <li>Tone Senders and Detectors</li> <li>Peripheral Equipment</li> <li>IP Network</li> </ul>	Insert (INST): IP Special Number (ISPN):
– Nodes: Servers, Media Cards – Maintenance and Reports – Media Gateways – Zones	Call Type to be used by the manipulated digits (CTYP) NPA (NPA)           Submit         Refresh         Delete         Cancel

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.** 

NØRTEL	CS 1000 ELEMENT MANAGER
- UCM Network Services - Home - Links - Virtual Terminals	Managing: <u>47.248.100.147</u> Username: admin Dialing and Numbering Plans » <u>Electronic Switched Network (ESN)</u> » Customer 00 » Network Control & Services » <u>Digit Manipulation</u>
- System	
+ Alarms - Maintenance	Input Description
- Core Equipment - Loops - Superloops - MSDL/MISP Cards - Conference/TDS/Multifrequen - Tone Senders and Detectors - Peripheral Equipment - IP Network - Nodes: Servers, Media Cards - Maintenance and Reports - Media Gateways	Digit Manipulation Index numbers (DMI): 25 Number of leading digits to be Deleted (DEL): 0 (0-19) Insert (INST): IP Special Number (ISPN): Call Type to be used by the manipulated digits (CTYP): NPA (NPA)
<ul> <li>Nodes: Servers, Média Cards</li> <li>Maintenance and Reports</li> <li>Media Gateways</li> <li>Zones</li> <li>Host and Route Tables</li> </ul>	Submit Cancel

Figure 47 – DMI Configuration Details Page.

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# 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)

NØRTEL	CS 1000 ELEMENT MANAGER
- UCM Network Services - Home - Links	Managing: 47.248.100.147 Username: admin Dialing and Numbering Plans » Electronic Switched Network (ESN) » C Doubted Light Plane Kee
- Virtual Terminals - System + Alarms - Maintenance - Core Equipment - Loops	Please enter a route list index 25 (0 - 999) to Add

Figure 49 – Add a Route List Blocks.

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

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

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



Figure 50 – Route List Blocks Configuration Details Page.

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

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

b) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen. Select **Route List Block** (RLB) as **Figure 48**.

c) Select an available value to and click **to Add** (in this case is 26)

NØRTEL	CS 1000 ELEMENT MANAGER	
- UCM Network Services - Home	Managing: <u>47.248.100.147</u> Username: admin Dialing and Numbering Plans » <u>Electronic Switched Network (ESN)</u> » Cus	
- Links - Virtual Terminals - System	Route List Blocks	
+ Alarms - Maintenance + Core Equipment	Please enter a route list index 26 (0 - 999) to Add	

Figure 51 – Add a Route List Blocks.

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

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



Figure 52 – Route List Blocks Configuration Details Page.

# 4.6.5. Inbound Call

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

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

b) Select **Dialing and Numbering Plans** -> **Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen. Select **Local Steering Code** (LSC).

NØRTEL	CS 1000 ELEMENT MANAGER
- UCM Network Services	Managing: <u>47.248.100.147</u> Username: admin Dialing and Numbering Plans » Electronic Switched Network (ESN)
- Virtual Terminals - System + Alarms	Electronic Switched Network (ESN)
+ Alarms - Maintenance - Core Equipment - Loops - Superloops - MSDL/MISP Cards - Conference/TDS/Multifrequen - Tone Senders and Detectors - Peripheral Equipment - IP Network - Nodes: Servers, Media Cards - Maintenance and Reports - Media Gateways - Zones - Host and Route Tables - Network Address Translation - QoS Thresholds - Personal Directories - Unicode Name Directory + Interfaces - Engineered Values + Emergency Services + Geographic Redundancy + Software Customere	- Customer 00 - Network Control & Services - Network Control Parameters (NCTL) - ESN Access Codes and Parameters (ESN) - Digit Manipulation Block (DGT) - Route List Block (RLB) - Incoming Trunk Group Exclusion (ITGE) - Network Attendant Services (NAS) - Coordinated Dialing Plan (CDP) - Local Steering Code (LSC) - Distant Steering Code (LSC) - Distant Steering Code (TSC) - Distant Steering Code (TSC) - Numbering Plan (NET) - Access Code 1 - Home Area Code (HNPA) - Home Location Code (HLOC) - Location Code (LOC) - Numbering Plan Area Code (NPA) - Exchange (Central Office) Code (NXX) - Special Number (SPN) - Network Speed Call Access Code (NSCL) - Free Calling Area Screening (FCAS)
- Customers - Routes and Trunks - Routes and Trunks - D-Channels - D-	<ul> <li>Free Special Number Screening (FSNS)</li> <li>Access Code 2</li> <li>Home Area Code (HNPA)</li> </ul>
Digital Trunk Interface     Digital Trunk Interface     Dialing and Numbering Plans     Electronic Switched Network     Flexible Code Restriction	<ul> <li>Home Location Code (HLOC)</li> <li>Location Code (LOC)</li> <li>Numbering Plan Area Code (NPA)</li> <li>Exchange (Central Office) Code (NXX)</li> </ul>

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** 

(	
NØRTEL	CS 1000 ELEMENT MANAGER
- UCM Network Services	Managing: <u>47.248.100.147</u> Username: admin Dialing and Numbering Plans » <u>Electronic Switched Network (ESN)</u> » Customer 00 » Coordinated Dialing Plan (CDP) » <u>Loc</u>
- Virtual Terminals	Local Steering Code
- System + Alarms - Maintenance	Input Description
- Core Equipment	Local Steering Code // SCh 121
- Loops	Local steering code (LSC). 151
- MSDL/MISP Cards	Digit Manipulation Index for LSC (DMI): 7 🔫
<ul> <li>Conference/TDS/Multifrequen</li> <li>Tone Senders and Detectors</li> </ul>	Number of digits to be deleted (DEL):
<ul> <li>Peripheral Equipment</li> <li>IP Network</li> <li>Nodes: Servers, Media Cards</li> <li>Maintenance and Reports</li> </ul>	Submit Cancel

Figure 55 – LSC Configuration Details Page.

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

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

a) Configure FCR in Customer by ld 15. This section prints FCR configuration details. >ld 21 PT1000

REQ: prt TYPE: fcr TYPE FCR\_DATA CUST 0 TYPE FCR\_DATA CUST 00 NFCR YES MAXT 100 OCB1 255 OCB2 255 OCB2 255 IDCA YES DCMX 100

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

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

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

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## 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).

NØRTEL	CS 1000 ELEMENT MANAGER
- UCM Network Services - Home - Links	Managing: <u>47.248.100.147</u> Username: admin Dialing and Numbering Plans » Electronic Switched Network (ESN)
- Virtual Terminals - System + Alarms	Electronic Switched Network (ESN)
<ul> <li>Maintenance</li> <li>+ Core Equipment</li> <li>- Peripheral Equipment</li> </ul>	- Customer 00     - Network Control & Services     - Network Control Parameters (NCTL)
+ IP Network + Interfaces - Engineered Values	<ul> <li>ESN Access Codes and Parameters (ESN)</li> <li>Digit Manipulation Block (DGT)</li> <li>Route List Block (RLB)</li> </ul>
+ Geographic Redundancy + Software	<ul> <li>Incoming Trunk Group Exclusion (ITGE)</li> <li>Network Attendant Services (NAS)</li> <li>Coordinated Dialing Plan (CDP)</li> </ul>
- Customers - Routes and Trunks - Routes and Trunks D. Chappede	<ul> <li>Local Steering Code (LSC)</li> <li>Distant Steering Code (DSC)</li> <li>Trunk Steering Code (DSC)</li> </ul>
- Digital Trunk Interface	- Numbering Plan (NET)
<ul> <li>Dialing and Numbering Plans         <ul> <li>Electronic Switched Network</li> <li>Flexible Code Restriction</li> <li>Incoming Digit Translation</li> </ul> </li> <li>Phones         <ul> <li>Templates</li> <li>Reports</li> <li>Properties</li> <li>Migration</li> </ul> </li> </ul>	<ul> <li>Access Code 1         <ul> <li>Home Area Code (HNPA)</li> <li>Home Location Code (HLOC)</li> <li>Location Code (LOC)</li> <li>Numbering Plan Area Code (NPA)</li> <li>Exchange (Central Office) Code (NXX)</li> <li>Special Number (SPN)</li> <li>Network Speed Call Access Code (NSCL)</li> <li>Free Calling Area Screening (FCAS)</li> <li>Free Special Number Screening (FSNS)</li> </ul> </li> </ul>

Figure 56 – ESN Configuration Page.

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

NØRTEL	CS 1000 ELEMENT MANAGER	
– UCM Network Services – Home – Links	Managing: <u>47.248.100.147</u> Username: admin Dialing and Numbering Plans » <u>Electronic Switched Network (ESN)</u> » Customer 00 :	
- Virtual Terminals - System + Alarms	Special Number List	
- Maintenance + Core Equipment	Please enter a Special Number 1 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)

NØRTEL	CS 1000 ELEMENT MANAGER	
- UCM Network Services - Home	Managing: <u>47.248.100.147</u> Username: admin Dialing and Numbering Plans » <u>Electronic Switched Network (ESN)</u> » Customer 00 » Numbering Plan (NET) > Access Code 1 » :	Special Number List » Special Number
- Virtual Terminals	Special Number	
- System + Alarms		
- Maintenance	Input Description	Input Value
- Peripheral Equipment	Special Number translation (SPN): 1	
+ IP Network + Interfaces	Flexible Length (FLEN): 11 (0-24)	
<ul> <li>Engineered Values</li> <li>Emergency Services</li> </ul>	- International Dialing Plan (INPL):	
+ Geographic Redundancy	Inhibit Time-out Handler (ITOH): 🕅	
+ Software - Customers	Route List Index (RLI): 25 🔻	
Routes and Trunks     Routes and Trunks     D-Channels     Digital Trunk Interface	Type of call that is defined by the special number (CLTP): No call type (NONE)	*
Dialing and Numbering Plans     Electronic Switched Network     Flexible Code Restriction     Incoming Digit Translation	Number to be Denied (DENY): (Items separated by a space)	*
- Phones - Templates	Digit Manipulation Index for LDID Numbers (DMI): 1 👻	
- Reports - Properties - Migration		×
- Tools + Backup and Restore - Call Server Initialization	- Local UN number to be recognized (LDIU): (Items separated by a space)	
- Date and Time		Ŧ

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"

NØRTEL	CS 1000 ELEMENT MANAGER
- UCM Network Services - Home	Managing: <u>47.248.100.147</u> Username: admin Dialing and Numbering Plans » <u>Electronic Switched Network (ESN)</u> » Customer 00 » Numbering Plan (NET) > .
- Virtual Terminals - System + Alarms	Special Number List
- Maintenance + Core Equipment	Please enter a Special Number 0 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) :
  - 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).
  - Select 25 (created in Section 4.6.3) to allow all phones able to make international call.

NØRTEL	CS 1000 ELEMENT MANAGER	
- UCM Network Services - Home	Managing: <u>47.248.100.147</u> Username: admin Dialing and Numbering Plans » <u>Electronic Switched Network (ESN)</u> » Customer 00 » Numbe	ring Plan (NET) > Access Code 1 » <u>Special Nun</u>
- Virtual Terminals	Special Number	
- System	epolaritanio i	
+ Alarms - Maintenance	Invest Descelation	
+ Core Equipment	Input Description	Input
- Peripheral Equipment	Special Number translation (SPN):	0
+ IP Network + Interfaces	Flexible Length (FLEN):	13 (0-24)
- Engineered Values	- International Dialing Plan (INPL):	
+ Emergency Services	Inhibit Time out Handler (ITOH)	
+ Software	innoit fine-out handler (from).	
- Customers	Route List Index (RLI):	26 -
<ul> <li>Routes and Trunks</li> <li>Routes and Trunks</li> <li>D-Channels</li> <li>Digital Trunk Interface</li> </ul>	Type of call that is defined by the special number (CLTP):	No call type (NONE)
Dialing and Numbering Plans     Electronic Switched Network     Flexible Code Restriction     Incoming Digit Translation	Number to be Denied (DENY): (Items separated by a space)	
- Phones - Templates	Digit Manipulation Index for LDID Numbers (DMI):	1 -

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

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

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

>ld 87 REQ chg CUST 0 FEAT nctl SOHQ OHTL SCBQ CBTL RANE RANC NCOS 7 EOA 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.

>Id 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

QT; Reviewed: SPOC 12/01/2010 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** 

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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 SL 1000 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

TN 960100 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.

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b. Select Configuration wizard	b.	Select	Configuratio	n Wizard
--------------------------------	----	--------	--------------	----------

CallPilot Manager - Home - Windows Internet Explorer			- 0 <b>- X</b> -
		- ft X Gauge	0 -
Tttp://47.246.100.90/cpmgf/default.asp		• • • • • • • • • • • • • • • • • • •	~ +
<u>File Edit View Favorites Tools H</u> elp			
👷 🔅 CallPilot Manager - Home		🟠 🔻 🔝 👻 🖶 👻 Page	▼ ③ Tools ▼ ″
N@RTEL CALLPILO	T MANAGER		良 î
LDAP server: 47.248.100.90   Mailbox Number: 000000		<u>Preferences</u>   <u>He</u>	lp   <u>Loqout</u>
Home User 🔻 System 👻 Maintenance 👻	Messaging 👻 Tools 🔻	Help 🔻	
Location + Home			
Home			
Welcome to CallPilot Manager			
Reset Password	Configuration Wizard		
Add User	Application Builder		
Download Player			
Copyright ⊜	2006 Nortel and its licensors. All rights	reserved.	-
	i 😜 Ii	nternet   Protected Mode: Off	🔍 100% 🔻 🔐

Figure 62 –Select Configuration Wizard.

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

CallPliot Manager - Configuration Wizard - Windows Internet Explorer
🚱 🔾 🝷 http://47.248.100.90/cpmgr/configwizard/ConfigMode.asp 🔹 4- 4- X Google 🖉
Eile Edit View Favorites Tools Help
😭 🏟 🌈 CallPilot Manager - Configuration Wizard 🏠 🔻 🔝 👻 🖶 Page 🕶 🍈 Tools 👻
NØRTEL CALLPILOT MANAGER
LDAP server: 47.248.100.90   Mailbox Number: 000000 Preferences   Help   Logout
Home User - System - Maintenance - Messaging - Tools - Help -
Location + Configuration Wizard + Configuration Mode
Configuration Wizard: Configuration Mode
Back Next Cancel Help
Configuration Mode:
<ul> <li>CallPilot System Configuration (Standard Mode)</li> </ul>
CallPilot Individual Feature Configuration (Express Mode)
Note: If you have a fresh installed system or a running system and want to do the system upgrade, please select the Standard Mode. With this selection you will be directed to go through the entire configuration procedures. However, if you currently have a running system and you want to reconfigure a specific feature, you may select the Express Mode.
Back Next Cancel Help

Figure 63 – Callpilot Configuration Details Page.

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

CallPilot Manager - Configuration Wizard - Windows Internet Explorer
Co v k http://47.248.100.90/cpmgr/configwizard/ExpressConfigList.asp v 4 X Google
File Edit View Favorites Tools Help
🔓 🏟 🖉 CallPilot Manager - Configuration Wizard
NØRTEL CALLPILOT MANAGER
LDAP server: 47.248.100.90   Mailbox Number: 000000 Preferences   Help   Logout
Home User 🔻 System 👻 Maintenance 👻 Messaging 💌 Tools 👻 Help 👻
Location + Configuration Wizard + Express Configuration List
Configuration Wizard: Express Configuration List
Express Configuration List:
Express Configuration Mode is designed for the individual feature(s) configuration. If you want to configure a fresh installed system or upgrade a currently running system, please go back to the previous page and select the Standard Configuration Mode, otherwise, you may continue the Express Configuration by selection the corresponding feature(c) from the following list.
Server Information
Media Allocation
Switch Configuration
Language Installation
Network Interface Card Configuration (ELAN and CLAN)
Back Next Cancel Help
Done Sinternet   Protected Mode: Off R 100% 🗸

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**.

communication betw	een ebrooo u	ia Cumpi	iot syste				
CallPilot Manager - Configuration W	/izard - Windows Internet Exp	lorer					
🚱 🔾 🗸 🙋 http://47.248.100.90/	cpmgr/configwizard/M1Swite	chInfo.asp		🕶 😽 🗙 Go	ogle	۶ -	
File Edit View Eavorites Tools	Help						
The fact from from the second se							
🙀 🐼 🌈 CallPilot Manager - Configuration Wizard 👘 🖈 💮 Page 🛪 💮 Tools 🛪							
Channel information for each Link	s displayed below. Click or	n a link to updat	e its channel se	ettings.		-	
OTI Depend 4 (2041) in plat 04)							
STI Board 1 (2011 In slot 01)							
Link STI01-001	Switch Type: 🧿 M1	1					
	⊙ M1	1 Option 11					
Link S1101-002	Switch Customer			Enable Syn	nposium Call Center S	Server	
	Number:			Integration			
	Switch IP Address: 47	248 1	00 147				
I							
						-	
	STI Board 201i		Boa	rd ID 68157440			
	Link STI01-001						
	# Channel Name +	TN	Key0	Key1	Channel Allocation		
	1 STI01-001-001	4.0.9.0	3201	3301	Multimedia		
	2 STI01-001-002	4.0.9.1	3202	3302	Multimedia		
	3 STI01-001-003	4.0.9.2	3203	3303	Multimedia		
	4 STI01-001-004	4.0.9.3	3204	3304	Multimedia		
	5 <u>STI01-001-005</u>	4.0.9.4	3205	3305	Multimedia		
	6 STI01-001-006	4.0.9.5	3206	3306	Multimedia		
	7 <u>STI01-001-007</u>	4.0.9.6	3207	3307	Multimedia		
	8 STI01-001-008	4.0.9.7	3208	3308	Multimedia		
	9 STI01-001-009	4.0.9.8	3209	3309	Multimedia		
	11 STI01-001-011	4.0.9.9	3210	3310	wuitimedia		
	12 STI01-001-012						
	40 OTIO4 004 042						
Done			😌 Interne	et   Protected Mod	e: Off	🔍 100% 🔻	

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** 

number would be CDIVeoning	ured on CD (refer to beetion 1.0.2) and check reek
CallPilot Manager - Configuration Wizard - Windows Ir	ternet Explorer
🕞 🔵 👻 🕖 http://47.248.100.90/cpmgr/configwiza	rd/M1CDNInfo.asp 🔹 🗸 🎸 🗙 Google 🔎
<u>File E</u> dit <u>V</u> iew F <u>a</u> vorites <u>T</u> ools <u>H</u> elp	
🖌 🎄 🌈 CallPilot Manager - Configuration Wizard	🟠 🔻 🗟 👻 🖶 Page 🕶 🎯 T <u>o</u> ols 🔻
LDAP server: 47.248.100.90   Mailbox Number: 000000	Preterences   Help   Loqout
Home User - System - Maintenance	▼ Messaging ▼ Tools ▼ Help ▼
Location → Configuration Wizard → Meridian 1 CDN Informatio	n tion
Back Next Cancel Help	
Meridian 1 CDN Information:	
Add new CDN entries corresponding to all those def	ined on the switch.
Note: The list of applications to be assigned is not o	complete, but create entries for all CDNs now because their initial definition requires a
reboot. Assignments can be changed later, and app	lications using other DNs can be added, using the Service Directory Number option under
the System menu.	
	ew Delete Selected
	# CDN Application Name
_	1 3111 Voice Messaging
	2 4300 Voice Messaging
	Delete Selected
Back Next Cancel Help	
Соругід	ht © 2006 Nortel and its licensors. All rights reserved.
Done	Internet   Protected Mode: Off

Figure 66 – Callpilot Configuration Details Page.

## g. Click Finish.

CallPilot Manager - Configuration Wizard - Windows Internet Explorer
🚱 🕞 👻 http://47.248.100.90/cpmgr/configwizard/ReadyToSubmit.asp 🔹 4 🗙 Google 🔎 🗸
<u>File E</u> dit <u>V</u> iew F <u>a</u> vorites <u>I</u> ools <u>H</u> elp
😭 🎄 🎉 CallPilot Manager - Configuration Wizard 🏠 🔻 🔂 👻 🖶 🖕 Page 🖛 🍈 Tools 🕶
NØRTEL CALLPILOT MANAGER
LDAP server: 47.248.100.90   Mailbox Number: 000000 Preferences   Help   Logout
Home User  System  Maintenance  Messaging  Tools  Help
Location + Configuration Wizard + Ready to Configure
Configuration Wizard: Ready to Configure
All the information required to complete the basic configuration of your CallPilot server has been entered. ≡
If you want to continue and apply the configuration to the CallPilot server, then click Finish to start the configuration process.
Note: Applying the configuration to the CallPilot server will not require any further action from you. However, the configuration process may take up to one hour to complete based on your server configuration.
Back Finish Cancel Help
Done Sinternet   Protected Mode: Off 🔍 100% 🔻

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 AACO NO RGAI NO ACAA NO

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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

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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

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

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

>ld 22 PT2000 REQ prt TYPE adan elan 16 ADAN ELAN 16 CTYP ELAN DES CPilot N1 512 REQ >ld 22 PT2000 REQ prt TYPE vas VAS VSID 016 DLOP ELAN 016 SECU YES **INTL 0001 MCNT 9999** 

#### e. Check the link between CS1000 and Callpilot

>ld 48 LNK000 .stat elan

SERVER TASK: ENABLED ELAN #: 016 DES: CPilot APPL\_IP\_ID: 47 .248 .100 .151 LYR7: ACTIVE EMPTY APPL ACTIVE

#### 4.9. CS1000 SIP-Line Configuration

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

#### 4.9.1. Configure SIP LINE CS1000 in Element Manager

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

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

- UCM Network Services	Managing: 47.248.10 System »	0.147 Username: ad IP Network » IP Telepho	imin ony Nodes			
- Home     - Links     - Virtual Terminals     Click the Node ID to view or edit its properties.     System						
+ Alarms - Maintenance	Add Import Export Delete					Print   Refre
+ Core Equipment	Node ID +	Components	Enabled Applications	ELAN IP	TLAN IP	<u>Status</u>
- IP Network	<u>1000</u>	1	LTPS, PD, Presence Publisher, Gateway ( SIPGw	-	47.248.100.244	Synchronized
- Maintenance and Reports	1001	1	SIP Line, LTPS, Gateway ( SIPGw	-	47.248.100.126	Synchronized
- maintenance and Reports						

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

# Figure 69 – Add a new node for SIP line.

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

NØRTEL	CS 1000 EL		IANAGER				
- UCM Network Services     - Home     - Links     - Virtual Terminals     - Virtual Terminals     - Virtual Terminals     - Varms     - Maintenance     + Core Equipment     - Peripheral Equipment     - IP Network     - Nodes: Servers. Media Cards     - Maintenance and Reports     - Maintenance and Reports     - Maintenance and Reports     - Media Gateways     - Zones     - Host and Route Tables     - Network Address Translation (N-     - QoS Translation (N-     - QoS Translation (N-     - QoS Translation (N-     - Cons     - Network Address     - Network     - Network Address     - Network     - Network	Managing: 47.248.100.147 User System » IP Network » Node Details (ID: 1002 Node ID: 1 Call Server IP Address: 4 Telephony LAN (TLAN) Node IP Address: 4 Subnet Mask: 2 IP Telephon • Voice Gateway (VGW • Outlivy of Sonice (ID)	1.248.100.147 Username: admin ystem »: P Network »: P Telephony Nodes tails (ID: 1002 - SIP Li) Node ID: 1002 * (0-9999) ver IP Address: 47.248.100.147 * hony LAN (TLAN) Del IP Address: 47.248.100.120 * Subnet Mask: 255.255.255.240 * IP Telephony Node Properties Cateway (VGW) and Codecs SIP Line Comparison of the second secon					E
- Unicode Name Directory + Interfaces - Engineered Values	LAN     * Required Value.			<ul> <li><u>Gateway</u></li> </ul>		Save	Cancel
Emergency Services     Geographic Redundancy     Software	Associated Signaling	Servers & Ca	rds				
- Customers	Select to add + Add	Remove	Make Leader			E	Print   Refresh
<ul> <li>Routes and Trunks</li> <li>Routes and Trunks</li> <li>D-Channels</li> </ul>	☐ Hostname ▲	Type Signaling Server	Deployed Application	<u>s</u>	ELAN IP	TLAN IP	Role
- Digital Trunk Interface     - Dialing and Numbering Plans     - Electronic Switched Network     - Flexible Code Restriction	Note: Only server(s) that are not available in the servers list .	part of any other IP to	slephony node and deploy	ed application(s	s) that match the service	ce(s) selected for this n	ode are

Figure 70 – Configure SIP line.

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

NØRTEL	CS 1000 EL	LEMENT MAN	AGER
- UCM Network Services - Home - Links - Virtual Terminals - System + Alarms - Maintenance + Core Equipment - Peripheral Equipment	Managing: 47.248.100.147 User System » IP Network » Node ID: 1002 - SIP Lin General   SIP Line Gateway SIP Li	name: admin <u>P Telephony Nodes</u> <b>ne Configuration D</b> o <u>Settings   SIP Line Gatewa</u> ine Gateway Application:	etails av Service I Enable gateway service on this Node
- IP Network	General		Virtual Trunk Network Health Monitor
<ul> <li>Nodes: Servers, Media Cards</li> <li>Maintenance and Reports</li> <li>Media Gateways</li> <li>Zones</li> <li>Host and Route Tables</li> <li>Network Address Translation (N. - QoS Thresholds</li> <li>Personal Directories</li> <li>- Unicode Name Directory</li> <li>Interfaces</li> <li>- Emgineered Values</li> <li>Emergency Services</li> </ul>	SIP Domain name	nterop.com         *           VRF14-SLS         1002           5070         (1 - 655)           5071         (1 - 655)	Monitor IP Addresses (listed below) Information will be captured for the IP addresses listed below. Monitor IP:Add Monitor addresses:  S5) Remove
+ Geographic Redundancy	SIP Line Gateway Settings		
+ Software - Customers - Routes and Trunks - Routes and Trunks - D-Channels - Digital Trunk Interface	Numbe	Security Policy: er of Byte Re-negotiation: Options:	Security Disabled   0  · Client Authentication  x509 Certificate Authentication Enabled  ·
<ul> <li>Dialing and Numbering Plans</li> <li>Electronic Switched Network</li> </ul>	* Required Value.	Note: Chai transmi	nges made on this page will NOT be Cancel Cancel

Figure 71 – SIP-Line Configuration Details Page.

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

NØRTEL	CS 1000 ELEMENT MANAGER	
- UCM Network Services - Home - Links - Virtual Terminals	Managing: 47.248.100.147 Username: admin System » P Network » <u>P Telephony Nodes</u> Node ID: 1002 - SIP Line Configuration Details	
- System	General   SIP Line Gateway Settings   SIP Line Gateway Service	
+ Alarms - Maintenance + Core Equipment	x509 Certificate Authentication Enabled	*
- Peripheral Equipment	SIP Line Gateway Service	
- IP Network     - <u>Nodes: Servers, Media Cards</u> Maintenance and Reports	SLG Role: MO	
- Maintenance and Reports - Media Gateways	SLG Mode: S1/S2 -	
- Zones - Host and Route Tables	MO SLG IP 47.248.100.120	
<ul> <li>Network Address Translation (N/ - QoS Thresholds</li> </ul>	MO SLG Port 5070 1 - 65535)	
<ul> <li>Personal Directories</li> <li>Unicode Name Directory</li> </ul>	MO SLG Transport: TCP 👻	
+ Interfaces	GR SLG IP: 0.0.0.0	
<ul> <li>Engineered Values</li> <li>Emergency Services</li> </ul>	GR SLG Port: 5070 (1 - 65535)	E
+ Geographic Redundancy + Software	GR SLG Transport: TCP 👻	
- Customers	IVR Settings:	
- Routes and Trunks	SLG IVR Proxy IP: 0.0.0.0	
- D-Channels - Digital Trunk Interface	SLG IVR Proxy Port: 5060 (1 - 65535)	+
- Dialing and Numbering Plans - Electronic Switched Network	* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved.	ave Cancel

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

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4. SIP\_LINE\_3P\_PKG - 416 - 3rdParty SIP Line Package

#### 4.9.3. Configure SIPL service in LD15

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

# 4.9.4. Configure DCH for SIPL in LD 17

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

#### 4.9.5. Configure ELAN AML link in LD 17

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

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#### 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 124000 - starting TN for virtual trunks

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

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

On Call Server >\*ld 96 DCH 011 : OPER EST ACTV AUTO DES: SIPL N1402 **On SLG** [nortel@vrf14-sls ~]\$ slgShow === VTRK === ===== General ====== SLG State = AppReady Total User Registered = 1 ===== AML Info ====== hAppBlk TaskName Tid LinkState NumRetry LinkNum Trace 0x1226c80 SLG 0xfb00 Up 0 32 0

#### 4.9.10. Setting password length for SIP line

LD 15 REQ CHG TYPE: **FFC** TYPE FFC\_DATA CUST 0

SCPL 4 – password length is 4

#### 4.9.11. Provisioning SIP client accounts on CS1000 Call Server

LD 11 REQ NEW TYPE UEXT

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

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SIPN 1 SIP3 1 FMCL 0 TLSV 0

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

\*\*\*End Note

SIPU **4197** – SIPL userID, often set equal to DN of the phone NDID **1002** – NodeID of the SIPL node ZONE 001 – MO zone configured in LD 117 TGAR **0** ... SCPW **1234** – password for SIPL client to log in ... CLS UNR

KEY 00 SCR **4197** – DN of the phone CPND **NEW** – in case you want to set CLID for phone NAME **set4197** XPLN 20 DISPLAY\_FMT FIRST,LAST 01 HOT U **2224197** 

#### 4.9.12. Check current status of set registration on SLG

[nortel@vrf14-sls ~]\$ slgSetShowAll

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# 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**.

Kortel Softphone 3456 Lo		
Username: Password:	nhan  Remember name Remember password Sign in automatically Forgot your password?	
		Sign in

Figure 73 – SMC3456 Log In.

# b) Click Skip

Ortel Softphone 3456 Lc	ogin	<u> </u>
Username:	nhan	
Password:	******	
	Remember name	
	Remember password	
	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**.

		<u> </u>
Application       Addition         Alerts & Sounds       Privacy         Quick Transfer       Devices         Devices       Network         Audio Codecs       Video Codecs         Video Codecs       Folder Locations         Quality of Service       LDAP         Contact Storage       Diagnostics         Advanced       Image: Contact Storage	dvanced         General Options         Image: Second Secon	

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**.

ccount Setti	ings					X
Enabled	Account name	Status	Protocol	User ID		Add >
	Account 2	Disabled	sip			Edit
						Remove
						Clear
						Clearm
						Move Up
						Move Down
The first enal	bled SIP account in the	he list is the de	fault account. Ph	one calls will be made on	this account if no	
and binn abbi						
					-	OK
					Apply	ОК

**Figure 77 – Accounting Settings** 

IP Account			
Account Voicemail	Topology Presence Storage	Security Advanced	
Account name:	4197	-	
Drotocok	cip.	-	
Protocol.	SIP		
User Details			
User ID:	4197@interop.com	e.g. joseph@domain.com	
Password:	****		
Display name:	4197		
Authorization name:	4197		
Send outbound via:			
Proxy Address: 4	47.248.100.120:5070		
Dial plan: #	#2\a\a.T;match=1;prestrip=2;		
		OK Cancel	

b) The created account is appeared as Figure 78.

**Figure 78 – SIP Account Detail Settings**
ccount Setti	ings					2
Enabled	Account name	Status	Protocol	User ID		Add >
<u>×</u>	4197	Ready	sip	4197@interop.com		Edit
					Press	mous
					K	inove
					-	Clear
					M	ove Un
					Mo	ve Down
"he firet ena	bled SID account in th	na liet ie tha d	efault account. Dh	one calle will be made on this a	ecount if no	
dial plan appl	lies to the dialed num	ber.	eraolt account. Ph	one cans will be made on this a	ocount in the	
					Apply	ок
					Арріу	UK

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.

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#### 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

REQ

OVLR NO OVLS NO

#### **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

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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 NDNO1\* DEXT NO DNAM NO ANTK SIGO STD STYP SDAT MFC NO ICIS YES

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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 50 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 00 FRL 10 FRL 20 FRL 30 FRL 40 FRL 50

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FRL 60 FRL 70 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 1000121 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

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

NACT

#### d. Administer SIP trunk gateway to SPS.

NØRTEL	CS 1000 ELEMENT MANAGER
- UCM Network Services - Home - Links - Virtual Terminals	Managing: 47.248.100.447 Username: admin System » IP Network » I <u>P Telephony Nodes</u> Node ID: 1001 - Virtual Trunk Gateway Configuration Details
- System + Alarms - Maintenance + Core Equipment - Peripheral Equipment - IP Network	General     SIP Gateway Settings   SIP Gateway Services       Vtrk Gateway Application:     ☑ Enable gateway service on this Node       General     Virtual Trunk Network Health Monitor
- Nodes: Servers, Media Cards     - Maintenance and Reports     - Media Gateways     - Zones     - Host and Route Tables     - Network Address Translation (N.     - Oos Thresholds     - Personal Directories     - Unicode Name Directory     + Interfaces     - Engineered Values     + Emergency Services     + Geographic Redundancy     + Software     - Customers	Vtrk Gateway Application:       SIP Gateway (SIPGw) •         SIP Domain name:       interop.com         Local SIP Port:       5060 • (1-6553)         Gateway endpoint name:       car1_ss2 • o         Gateway password:       *         Enable failsafe NRS:       Remove
- Routes and Trunks - Routes and Trunks - D-Channels - Digital Trunk Interface - Dialing and Numbering Plans - Electronic Switched Network	SIP Gateway Settings TLS Security: Security Disabled  Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved. Save Cancel

Figure 81 – Virtual trunk gateway Configuration Details.



Figure 82 – Virtual Trunk Gateway Configuration Details.

NØRTEL	CS 1000 ELEM	MENT MANAGE	R		
- UCM Network Services	Managing: 47.248.100.147 Username System » IP Network » IP Te	e: admin Hephony Nodes			
- Links	Node ID: 1001 - Virtual Tru	unk Gateway Configu	ration Details		
- Virtual Terminals					
- System	General   SIP Gateway Settings	SIP Gateway Services			
+ Alarms			· · · · · · · · · · · · · · · · · · ·	-,	
- Maintenance		Subscriber (SN): 0	<ccc>&lt;</ccc>	Area code> <sn></sn>	<u>^</u>
+ Core Equipment		National (NN): 0		AININ	
- Peripheral Equipment		National (NN). 0	<0002	1414 >	
- IP Network		International: 0	<internat< td=""><td>ional number&gt;</td><td></td></internat<>	ional number>	
Maintenance and Reports     Media Gateways     Zones     Host and Route Tables     Host and Route Tables     Network Address Translation (Ni     OoS Thresholds     Personal Directories     Unicode Name Directory     Interfaces     Emgineered Values     Emgrency Services     Geographic Redundancy     Software	SIP URI Map: Public E. 164 D National. + Subscriber: Special number: F Unknown: F	omain Names 1 JublicSpecial JublicUnknown	P Special Vacant U	rivate Domain Names UDP: udp CDP: cdp.udp number: PrivateSpecial number: PrivateUnknown Inknown: UnknownUnknown	ш
- Customers	SIP Gateway Services				
- Routes and Trunks	SIP Converged Desktop:	Enable CD service			
- D-Channels - Digital Trunk Interface	The second second second second	Service DN:	Used for making	3 VTRK call from agent.	-
- Dialing and Numbering Plans - Electronic Switched Network	* Required Value.	Note: Changes ma transmitted until	de on this page will NOT be the Node is also saved.	Save	ancel

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

NØRTEL	CS 1000 ELEMENT MANAGER	
- UCM Network Services - Home - Links	Managing: <u>47,248.100.147</u> Username: admin Dialing and Numbering Plans » <u>Electronic Switched Network (ESN)</u> » Customer 00 » Network C	ontrol & Services » <u>Digit Manipulation Block List</u> » Digit Manipulat
- Virtual Terminals - System + Alarms	Digit Manipulation Block	
- Maintenance	Input Description	Input Value
- Peripheral Equipment     - Peripheral Equipment     + IP Network     + Interfaces     - Engineered Values	Digit Manipulation Index numbers (DMI): 50 Number of leading digits to be Deleted (DEL): 0	(0-19)
+ Emergency Services	Insert (INST):	
+ Geographic Redundancy	IP Special Number (ISPN):	
- Customers	Call Type to be used by the manipulated digits (CTYP): Co	oordinated Dialing Plan (CDP) 🔹
<ul> <li>Routes and Trunks</li> <li>Routes and Trunks</li> <li>D-Channels</li> </ul>	Submit Refresh Delete Cancel	



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



Figure 85 – Route List Blocks Configuration Details.

c) Create Distant Steering Code (DSC) to route the call to CS1000\_B.

NØRTEL	CS 1000 ELEMENT MANAGER	
- UCM Network Services - Home - Links - Virtual Terminals - System	Managing: <u>47.248.100.147</u> Username: admin Dialing and Numbering Plans > <u>Electronic Switched Network (ESN)</u> > Customer 00 > Coordin Distant Steering Code	ated Dialing Plan (CDP) » <u>Distant Steering Code List</u> » Distant Steering
+ Alarms - Maintenance + Core Equipment	Input Description	Input Value
<ul> <li>Peripheral Equipment</li> <li>IP Network</li> <li>Interfaces</li> </ul>	Distant Steering Code (DSC): Flexible Length number of digits (FLEN):	50 6 (0-10)
- Engineered Values + Emergency Services + Geographic Redundancy	Display (DSP): Remote Radio Paging Access (RRPA):	Local Steering Code (LSC)
- Customers	Route List to be accessed for trunk steering code (RLI):	50 💌
- Routes and Trunks	Collect Call Blocking (CCBA):	
- D-Channels	maximum 7 digit NPA code allowed (NPA):	
- Dialing and Numbering Plans     - Electronic Switched Network	maximum 7 digit NXX code allowed (NXX):	
- Flexible Code Restriction - Incoming Digit Translation	Submit Refresh Delete Cancel	

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#### 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.

NØRTEL	CS 1000 ELEMENT MANAGER	
- UCM Network Services     - Home     - Links     - Virtual Terminals     - System     + Alarms	Managing: <u>47.248.100.147</u> Username: admin Dialog and Numbering Plans » <u>Electronic Switched Network (ESN)</u> » Customer 00 » Network Control & Services » <u>Diat Manipulation Block List</u> » Digit Ma Digit Manipulation Block	inipulatic
Maintenance     Core Equipment     Peripheral Equipment     IP Network     Interfaces     Engineerad Values     Emergency Services     Geographic Redundancy     Software	Input Description Input Value Digit Manipulation Index numbers (DMI): Number of leading digits to be Deleted (DEL): Insert (INST): IP Special Number (ISPN):	
- Customers - Routes and Trunks - Routes and Trunks - D-Channels - Digital Trunk Interface	Call Type to be used by the manipulated digits (CTYP): Coordinated Dialing Plan (CDP)	

Figure 87 – DMI Configuration Details.

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

NØRTEL	CS 1000 ELEMENT MANAGER		
- UCM Network Services - Home	Managing: 47.248.100.147 Username: admin Dialing and Numbering Plans » <u>Electronic Switched Network (ESN)</u> » Customer 00 » Coordinated Dialing Plan (CDP) » <u>Local Stee</u>		
- Virtual Terminals	Local Steering Code		
- System + Alarms - Maintenance	Input Description		
- Peripheral Equipment	Local Steering Code (LSC): 53		
+ IP Network + Interfaces Digit Manipulation Index for LSC (DMI): 2 -			
- Engineered Values     + Emergency Services     + Geographic Redundancy     + Software     - Customers	Number of digits to be deleted (DEL):     (1-7)       Submit     Refresh     Delete		

Figure 88 – Local Steering Code Configuration Details.

#### 4.10.2.5 Configure Dialing Plan - route a call from PSTN to CS1000\_B

When there is a call from PSTN to the online number 12107574598, this call will come to CS1000\_A first. If user wants to receive this call by a phone on CS1000\_B, user has to configure on CS1000\_A to forward it to CS1000\_B. To re-route a call from CS1000\_A to CS1000\_B, user can use IDC feature as follows.

a) Configure FCR in Customer by ld 15. This section prints FCR configuration details. >ld 21 PT1000

REQ: prt TYPE: fcr

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TYPE FCR\_DATA CUST 0

TYPE FCR\_DATA CUST 00 NFCR **YES** MAXT **100** OCB1 255 OCB2 255 OCB3 255 IDCA **YES** DCMX **100** 

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

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

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

LD 16 ROUT 100 IDC YES DCNO 1 NDNO 1 \* \*\*\*\*\*\*\*\*End Note \*\*\*\*\*\*\*\* SDID NO IDGT CDGT 12107574698 504698 13157914457 3111

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

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

NØRTEL	CS 1000 ELEMENT MANAGER
- UCM Network Services - Home Links	Managing: 47.248.100.147 Dialing and Numbering Plans » Incoming Digit Translation » Customer 00 » Digit Conversion Tree 1 Configuration
- Virtual Terminals - System + Alarms	Digit Conversion Tree 1 Configuration Regular IDC tree
- Maintenance     + Core Equipment     - Peripheral Equipment     HP Network	Send calling party DID disabled
+ Interfaces - Engineered Values + Emergency Services	Add     Delete IDC     Delete IDC tree       Incoming Digits +     Converted Digits     CPND Name
+ Geographic Redundancy + Software - Customers	1 (a)         12107574698         504698           2 (b)         13157914457         3111
- Routes and Trunks     - Routes and Trunks     - D-Channels     - Digital Trunk Interface	
- Dialing and Numbering Plans     - Electronic Switched Network     - Flexible Code Restriction	
- Incoming Digit Translation	

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

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

REQ

#### **b.** Administer Virtual SIP Routes

>ld 21 PT1000 REQ: prt TYPE: rdb CUST 0 **ROUT 101** TYPE RDB CUST 00 **ROUT 101** DES ENTERPRISE TKTP TIE M911P NO ESN NO RPA NO **CNVT NO** SAT NO **IDEF NET** RCLS EXT VTRK YES **ZONE 255** PCID SIP CRID NO **NODE 2001** DTRK NO ISDN YES MODE ISLD DCH 101 IFC SL1 PNI 00001 NCNA YES NCRD YES TRO NO FALT NO CTYP UKWN INAC YES ISAR NO DAPC NO

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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

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EESD 1024 SST 50 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 00 FRL 10 FRL 20 FRL 30 FRL 40 FRL 50 FRL 60 FRL 70 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 100010 DATE PAGE DES ENTER TN 10000100 VIRTUAL TYPE IPTI CDEN 8D CUST 0 XTRK VTRK **ZONE 255** LDOP BOP **TIMP 600 BIMP 600** AUTO\_BIMP NO NMUS NO TRK ANLG NCOS 7 RTMB 101 1 CHID 1 TGAR 0 STRI/STRO IMM IMM SUPN YES AST NO IAPG 0 CLS UNR DTN CND ECD WTA LPR APN THFD XREP SPCD MSNV P10 NTC MID TKID AACR NO DATE 27 AUG 2010

#### NACT

#### d. Administer SIP trunk gateway to SPS.

Basing on network topology shown in Section 4.10.1, below is the configuration details of SIP trunk on CS1000\_B to SPS.

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NØRTEL	CS 1000 ELEMENT MANAGER	
- UCM Network Services	Managing: 47.248.100.138 Username: admin System » P Network » P Telephony Nodes	
- Home - Links - Virtual Terminals	Node ID: 2001 - Virtual Trunk Gateway Configuration Details	_
- System	General I SIP Gateway Settings I SIP Gateway Services	
+ Alarms	General VIITUAL I FUNK NETWORK HEAITIN MONITOR	
<ul> <li>Maintenance</li> <li>+ Core Equipment</li> </ul>	Vtrk Gateway Application: SIP Gateway (SIPGw) 🔹 🥅 Monitor IP Addresses (listed below)	
Peripheral Equipment     IP Network     Nodes: Servers, Nedia Cards	SIP Domain name: interop.com Information will be captured for the IP addresses listed below.	H
- Maintenance and Reports	Local SIP Port: 5060 * (1 - 65535) Monitor IP: Add	
- Zones - Host and Route Tables	Gateway endpoint name: car2_ss2 * Monitor addresses:	
<ul> <li>Network Address Translation (N/ - QoS Thresholds</li> <li>Personal Directories</li> </ul>	Gateway password:	
- Unicode Name Directory	Enable failsafe NRS:	
- Engineered Values		
+ Geographic Redundancy	SIP Gateway Settings	
- Customers	TLS Security: Security Disabled -	
- Routes and Trunks	Port: 5061 (1 - 65535)	
<ul> <li>Routes and Trunks</li> <li>D-Channels</li> </ul>	Number of Bute Re-negotiation	
- Digital Trunk Interface	Humber of Byte Nonegotiution.	*
- Dialing and Numbering Plans - Electronic Switched Network	* 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.



**Figure 91 – Virtual trunk gateway configuration details** 

UCM Network Services Home Links	Managing: 47.248.100.138 Username: admin System » IP Network » <u>IP Telephony Nodes</u> Node ID: 2001 - Virtual Trunk Gateway Configura	tion Details
- Virtual Terminals System	General I SIP Gateway Settings   SIP Gateway Services	
+ Alarms - Maintenance + Core Equipment	International: 0	<pre></pre>
- Peripheral Equipment - IP Network	SIP URI Map:	
- Nodes: Servers, Media Cards	Public E.164 Domain Names	Private Domain Names
<ul> <li>Maintenance and Reports</li> <li>Media Cateways</li> </ul>	National: +1	UDP: udp
- Zones	Subscriber:	CDP: cdp.udp
<ul> <li>Host and Route Tables</li> <li>Network Address Translation (N/</li> </ul>	Special number: PublicSpecial	Special number: PrivateSpecial
<ul> <li>QoS Thresholds</li> <li>Personal Directories</li> </ul>	Unknown: PublicUnknown	Vacant number: PrivateUnknown
- Unicode Name Directory Interfaces		Unknown: UnknownUnknown
- Engineered Values - Emergency Services	SIP Gateway Services	
+ Geographic Redundancy	SIP Converged Desktop: Enable CD service	
+ Software Customers	Service DN:	Used for making VTRK call from agent.
Routes and Trunks	Converged telephone call forward DN:	
- Routes and Trunks - D-Channels	RAN route for Announce:	(route number 0 - 511)

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

LQ pu

ESN004 REQ prt CUST 0 FEAT rlb RLI 53

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**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

CBQ NO

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

QT; Reviewed: SPOC 12/01/2010 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
----	-----------	---------	----------	---------	------------	---------

- Network Elements	Host Name: nd2-carrier2.interop.com Softwar	re Version: 02.00.0055.00(3266)	User Name admin			
- CS 1000 Services	Elements					
Patches SNMP Profiles	New elements are registered into the security framework, or may be added as simple hyperlinks. Click an element name to launch its management service.					
Secure FTP Token Software Deployment	Add Edit Delete					
Subscriber Manager	Element Name	Element Type -	Release	Address		
- User Services Administrative Users	1 EM on nd2-carrier2	CS1000	6.0	47.248.100.138		
External Authentication Password	2 EM on nd2-ss	CS1000	6.0	47.248.100.138		
Roles	3 🗐 sipl-6.interop.com (member)	Linux Base	6.0	47.248.100.131		
Policies Certificates	4 md2-ss.interop.com (member)	Linux Base	6.0	47.248.100.253		
Active Sessions	5 md2-carrier2.interop.com (primary)	Linux Base	6.0	47.248.100.251		
Logs	e 📄 sps3.interop.com (member)	Linux Base	6.0	47.248.100.150		
	7 🔲 47.248.100.137	Media Gateway Controller	6.0	47.248.100.137		
	s 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.

N@RTEL	NETWORK ROUTING SERV	ICE MANAGER		
«UCM Network Services - System NRS Server Database	Managing: O Active database Standby database	47.248.100.150 <u>Numbering Plans</u> ,» Domains		
System Wide Settings - Numbering Plans Domains Endpoints Routes Network Post-Translation CollaborPative Servers	Domains Domains establish the basic structure of you Service Domains (1)	r converged network, defined by Service Domains (UDP) (1) L0 Do	domains, L1 (UDP) and L0 (CD omains (CDP) (1)	P) domains.
- Tools SIP Phone Context - Routing Tests H.323 SIP Backup Restore GK/NRS Data upgrade	Domain Name •	Description Interop.com	# of L1 Domains	# of L0 Domains

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

NØRTEL N	IETWORK ROUTIN	G SERVICE MAN	AGER				
«UCM Network Services - System NRS Server Database	Managing: CACtive database 47.248.100.150 Standby database Numbering Plans, » Endpoints						
System Wide Settings - Numbering Plans Domains	Search for Endpoint	ints					
Endpoints Routes Network Post-Translation Collaborative Servers - Tools SIP Phone Context - Bouting Tests	Enter an endpoint ID (use * Endpoint ID: * Limit results to Domain: in	for all) and click Search.You may terop.com	narrow the search by s	pecifying a particular domain	n.		
H.323 SIP						Results per	
Backup Restore	Gateway Endpoints	(6) User Endpoint	s (1)				
GK/NRS Data upgrade	Add Delete	SIP phone context					
		Supported Protocols	SIP Mode	Call Signaling IP	Description	# of Routing Entries	
	1 📃 <u>car1 ss2</u>	Dynamic SIP endpoint / NCS	Proxy Mode	Not available	car1_ss2	8	
	2 🔲 <u>car2-ss3</u>	Dynamic SIP endpoint / NCS	Proxy Mode	Not available	car2-ss3	0	
	3 🔲 <u>car2_ss2</u>	Dynamic SIP endpoint	Proxy Mode	Not available	car2_ss2	5	
	4 🥅 <u>mp118 1</u>	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

NØRTEL NET	IWORK ROUTING SERVICE MANAGER
«UCM Network Services - System NRS Server Database System Wide Settings - Numbering Plans	Managing:         O Active database         47.248.100.150           Image: Standby database         Numbering Plans.» Endpoints.» Qate way Endpoint           Edit Gateway Endpoint ( interop.com / udp / cdp )
Domains Endpoints	End point name: car1_ss2 *
Routes Network Post-Translation Collaborative Servers	Description:
- Tools	Trust Node: 📝
SIP Phone Context	Tandem gateway endpoint name: Not Applicable 👻
H.323	Endpoint authentication enabled: Authentication off 🝷
SIP Backup	Authentication password:
Restore GK/NRS Data upgrade	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** 

NORTEL	NETWORK ROUTING SERVICE MANAGER	Heir
<ul> <li>UCM Network Services</li> <li>System</li> <li>NRS Server</li> <li>Database</li> <li>System Wide Settings</li> </ul>	Managing: C Active database 47.245.100.150 Standby database <u>Numbering Plans, » Endoptins, » Gateway Endoptin</u> Edit Gateway Endpoint ( interop.com / udp / cdp )	
Numbering Plans     Domains     Endpoints     Routes     Network Post-Translation     Collaborative Servers     Toolis     SIP Phone Context     SIP Phone Context     Routing Tests	Static endpoint address type: IP virriant 4 • Static endpoint address: H 323 support H 323 net supported • SIP support Dynamic SIP endpoint • SIP Mode • SIP Mode •	
H 323 SIP Backup Restore GKINRS Data upprade	SIP TCP transport enabled SIP TCP port 5060 SIP UDP transport enabled SIP UDP port 5060 SIP UDP port 5060 SIP TLS transport enabled SIP TLS transport enabled SIP TLS transport enabled SIP TLS port 5061 Persistent TCP support enabled SIP End to end security support SIP	
	* Required value	Save

**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.

NORTEL N	ETWORK ROUTING S	SERVICE MANA	AGER			
«UCM Network Services - System NRS Server Database	Managing:      Active database     Standby database	47.248. se Number	100.150 ing Plans » Endpoints			
System Wide Settings - Numbering Plans Domains	Search for Endpoints					
Endpoints Routes Network Post-Translation Collaborative Servers <b>Toots</b> SIP Phone Context <b>Routing Tests</b> H 323 SIP Backup Restore GKINRS Data upgrade	Enter an endpoint ID (use * for all Endpoint ID: Car* Limit results to Domain: interop	) and click Search.You may r	arrow the search by spe	cifying a particular domain.		Results ner nar
	Gateway Endpoints (3) SIP phone context	User Endpoints	(0)			
		Supported Protocols Dynamic SIP endpoint /	SIP Mode	Call Signaling IP	Description	# of Routing Entries
	2 Car2-ss3	NCS Dynamic SIP endpoint / NCS	Proxy Mode	Not registered	car2-ss3	0
	3 <u>car2 ss2</u>	Dynamic SIP endpoint	Proxy Mode	47.248.100.252	car2_ss2	5

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.

NØRTEL N	ETWORK ROUTI	NG SERVICE MANAGE	IR		l
«UCM Network Services - System NRS Server Database	Managing: O Active	database 47.248.100.11 y database Numbering Pic	i0 ns_» Routes		
System Wide Settings - Numbering Plans Domains	Search for Routing	) Entries			
Endpoints Routes Network Post-Translation	Enter a DnPrefix and Dn	Type (use * for all) and click Search.You m	ay narrow the search by sp regional (CDP steering c	ecifying a particular domain. ode) 🔻	
- Tools SIP Phone Context - Routing Tests H.323 SIP	Limit results to Domain:	interop.com  v / udp Name: Car1_ss2 v	▼ / cdp	•	
Backup Restore					Results per page: 50 👻
GK/NRS Data upgrade	Routing Entries	(5) Default Routes (0)			
	Add Copy	Move Import Export Rout	ing test Delete		0.111
	1 1 31	DN Type Private level 0 regional (CDP steering code)	1	cdp.udp	interop.com / udp / cdp / car1_s
	2 🛄 32	Private level 0 regional (CDP steering code)	1	cdp.udp	interop.com / udp / cdp / car1_s
	3 🗐 43	Private level 0 regional (CDP steering code)	1	cdp.udp	interop.com / udp / cdp / car1_s
	1 - 5 of 5 Routing Entry(ies)		Page 1 of	'1	First  Pre

Figure 99 – Routes Configuration Page.

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

NØRTEL	ETWORK R	OUTING SERVI	CE MANAGER	
«UCM Network Services - System NRS Server	Managing:	Active database Standby database	47.248.100.150 Numbering Plana.» Routes.» Routing Entry	
System Wide Settings	Edit Routing	Entry ( interop.com /	udp/cdp/car1_ss2)	
Numbering Plans	1. 2. 2. 2. 2. 2. 2. 2. 2. 2. 2. 2. 2. 2.			
Domains			DN type: Private level 0 regional (CDP steering code) -	
Endpoints			ON state 12	
Routes			Div prenx: 53 *	
Network Post-Translation			Route cost. 1 * (1-255)	
Collaborative Servers				
DID Dhana Context				
- Pouting Tosts				
H 323	* Required value.			Save
SIP				
Backup				
Restore				
GK/NRS Data upgrade				

Figure 100 – CS1000\_A (car1\_ss2) - Routes Configuration Details Page.

c) Do the same to create NPA 1613.

«UCM Network Services - System NRS Server	Managing:	<ul> <li>Active database</li> <li>Standby database</li> </ul>	47.248.100.160 Numbering Plans, » Routes, » Routing Entry		
Database System Wide Settings Numbering Plans	Edit Routi	ng Entry ( interop.com )	udp/cdp/car1_ss2)		
Domains Endpoints Routes Network Post-Translation Collaborative Servers Tools			DN type: E 164 national DN prefix 1613 Route cost 1 (1-255)	-	
SIP Phone Context Routing Tests H.323 SIP Backup Restore	* Required va	lue.			Sav

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**.

NORTEL	NETWORK ROUTING SERVICE MANAGER	Helo
«UCM Network Services - System NRS Server Database	Managing: Active dalabase 47.248.100.150 Standby dalabase <u>Humberino Pans</u> - Routes	
System Wide Settings - Numbering Plans Domains	Search for Routing Entries	
Endpoints Routes	Enter a DnPrefix and Dn Type (use * for all) and click Search. You may narrow the search by specifying a particular domain.	
Collaborative Servers	DN Prefix * DN Types *	
Tools		
SIP Phone Context	Limit results to Domain interop com 🔹 / udp 👻 / cdp 👻	
- Routing Tests H.323 SIP	Endpoint Name, car1_ss2 •	
Backup		Results per page: 50 - 5
Restore		
Giorino Dala upgrade	Routing Entries (2) Default Routes (0)	
	Add Copy Move Import Export Routing test Delete	
	DN Pretix - DN Type Route Cost SIP URI Phone Context	Context
	1 1613 E 164 national 1 +1	interop.com / udp / cdp / car1_ss2
	2 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**.

NØRTEL N	ETWORK ROUTING	SERVICE MANAG	ER		Help
«UCM Network Services - System NRS Server Database	Managing: O Active datab Standby data	ase 47.248.100. Ibase Numbering F	150 Tans = Routes		
System Wide Settings - Numbering Plans Domains	Search for Routing En	tries			
Endpoints Routes	Enter a DnPrefix and Dn Type	use * for all) and click Search.You r	may narrow the search by spec	ifying a particular domain.	
Network Post-Translation Collaborative Servers	DN Prefix	DN Type All DN Type		-	
- Tools	-				
SIP Phone Context	Limit results to Domain; inter	op.com 👻 / udp		•	
- Routing Tests H 323					
SIP	Endpoint Nam	e: All gateway endpoints 👻			
Backup Restore					Results per page: 50 👻
GK/NRS Data upgrade	Routing Entries (5)	Default Routes (0)			
	mercani increase (c)		Investorial Investorial		
	Copy Inc	Import Export Roo	uting test Delete		
	DN Prefix +	DN Type	Route Cost	SIP URI Phone Context	Context
	1 1613 E.1	64 national	1	+1	interop.com / udp / cdp / car1_ss2
	2 50 Pri	ate level 0 regional (CDP steering	1	cdp.udp	interop.com / udp / cdp/ car2_ss2
	3 53 Pri	ate level 0 regional (CDP steering	1	cdp.udp	interop.com / udp / cdp / car1_ss2
	613 F1	e) 64 national	4	+1	interon.com / udn / cdn / node3-300
	1 - 5 of 5 Routing Entry(ies)		Page 1 of 1		First  Previous

Figure 103 – Routes Entries for CS1000\_A (car1\_ss2).

### 4.10.4.3 Save Configuration

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

	, O
NØRTEL	NETWORK ROUTING SERVICE MANAGER
«UCM Network Services     System     NNS Server     Catabase     System Vide Settings     System Vide Settings     Ormains     Domains     Domains     Routes     Network Post-Translation     Collaborative Servers     SizP Phone Context     SuiP Phone Context     H 323     Siz	Managing:       47.246.100.150 System × Database         Database       Database         NRS uses a redundant database with Active and Standby copies. Normally changes are made to the standby database, tested, then cut over into active status.         Database status: Changed       Cut over I
SIP Phone Context - Routing Tests H.323 SIP	

#### Figure 104 – Database Cut Over.

b) Click on Commit as shown in Figure 105.

N@RTEL	NETWORK F	ROUTING SERVICE MANAGER		
«UCM Network Services - System	Managing:	47.243.100.150 System a Database		
Database System Wide Settings	Database NRS uses a rec	dundant database with Active and Standby copies. Normally changes are made to the standby database, tested, then cut ove	r into active st	latus.
- Numbering Plans Domains Endpoints	Database stat	tus: Switched over	Revert	Commit
Routes Network Post-Translation Collaborative Servers				

Figure 105 – Database Commit.

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

NØRTEL	NETWORK ROUTING	SERVICE MAN	AGER			
«UCM Network Services – System NRS Server Database	Managing:   Active databation  Standby data	se 47.24 base <u>Numb</u>	3.100.150 ering Plans » Endpoints			
System Wide Settings     Numbering Plans     Demoins	Search for Endpoints					
Domains Endpoints Routes Network Post-Translation Collaborative Servers - Tools SIP Phone Context - Routing Tests H 323	Enter an endpoint ID (use * for Endpoint ID: * Limit results to Domain: All so	all) and click Search.You may	narrow the search by s domains  ▼ / All Lt	pecifying a particular domain 0 domains 👻	l	Results per p
Backup Restore	Gateway Endpoints (7)	User Endpoint	s (1)			
GK/NRS Data upgrade	SIP phone context					
	<u>□</u> <u>□</u> •	Supported Protocols	SIP Mode	Call Signaling IP	Description	# of Routing Entries
	1 📰 <u>car1 ss2</u>	Dynamic SIP endpoint / NCS	Proxy Mode	47.248.100.126	car1_ss2	8
	2 🔲 <u>car2-ss3</u>	Dynamic SIP endpoint / NCS	Proxy Mode	Not registered	car2-ss3	0
	3 Car2 ss2	Dynamic SIP endpoint	Proxy Mode	47.248.100.252	car2_ss2	<u>5</u>
	4 🥅 <u>mp118 1</u>	Dynamic SIP endpoint	Proxy Mode	Not registered	mp118_1	1
	5 mp118 mcs usr	Dynamic SIP endpoint	Proxy Mode	Not registered	mp118_mcs_us	r <u>1</u>
	1 - 7 of 7 Gateway Endpoint(s)			Page 1 of 1		

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 "<u>http://download.skype.com/share/business/guides/skype-for-sip-user-guide.pdf</u>"

# 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 <u>skype.com/business</u> 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

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

Sign in to your Skype account - Microsoft Inter	met Explorer		
Mie Edit Weve Parvantes Toole Help	the state of the second		
3 Back + 3 - 💌 🖉 🏠 🔎 Search	👷 Pausetas 🙆 🍰	• 🍓 💷 - 🛄 🛍	
Addinan 🍓 https://login.skype.com/bcp/login/nessage=logi	teriuper_n		
Skype		Buy Stype Credit + Ab	nusdy have Skype? • Help • Search
Download	Use Skype	Susiness Shop Account	
Fastures	Producte Solutions	Case studies Partners Support	Skype Menager
Sign in			Don't have a Skype Manager yet?
You need to sign in with you	ur Słape Name to continue.		Desister a Flase Manager
Skype Name		Forgotten your Skype Nam	now and set up, manage and monitor the use of Sitype in your company.
Password		Forgotten your password?	Set up now
		Sign me In	

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

🖥 Dashkumit - Skype Man	awyer - Skype			<b>₫・日・□</b> @	• Rope • Selety • Tools • 😜
Skype m	anager" Step 1: click on Feature			Anigra - Account Onlaria -	langahyaatt - Bigraw - Hilo
Castioant	4 (29) B	9	€73,29	Buy Skype Credit	Q, Inscontern
	Reports	Your account	Ab	count status: Attaintion requ	not de
	Allocations	Current balance 😡	191		
	204		€73,29	O Ado rethings 6 d	housed
		O (Devi 30 dans)	€120,60	Radow at:	-Catana
			Credit miss	ing €47,30	
	the set of eq be to be to be a fee to be by		Buy new 3	Skype Credit	
	Your features	embers	Ner	ws	
	Streenbers have Swore Create Visur Disper Manager has	Tirenses	Welkernet Single	Marsacor Io. a. brand their and	10.

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.

🔁 Profile 1 - Skype for SP - Skype Mana	iger - Skype		🟠 🔹 🔯 🕆 📑 🖶 🔹 Page 🕶 Safety 🕶 Tools
	Profile settings		
Profile 1			200
	Profile name	Profile 1	
Profile settings	Calling channels	6 chainea 🖸	
Authentication details	Outgoing calls	€37,54 Auto-recharge active	
Reports	Callier ID 😡	Caller ID is set to 🐜 =16174018237	
• Back to SPI Profile Uni	incoming calls	+16174018237	
		athampt102	
		Extension number (optional)	
		1874 Save Settings	
		View account details	
		Remove account	
		Add a number of business account	

Figure 111 – Skype Profile Settings

skype manager				Avaga - Account details	- tony.skype11 - Sign.out - He
🙆 🤐 👰 Features	<u>ا</u>		€73,29	Buy Skype Credit	Q, Search Members
B	Authentication details				.0
Profile 1	Please choose the method	of authentication needed for your PBX.			
Profile settings	P. Charles				
Authentication details	<ul> <li>Registration (Usemamelpessword)</li> </ul>	or, IP Authentication 😜			
Reports	SIP User	94349764676146			
Particle CED Participation	Password	w5NUDpxhas494I Generale a new password			
e Back to SP Prome list	Skype for SIP address	sip.skype.com			
	UDP Port	5050			
	SIP user successfully registere Last registration: May 25, 2010	d at sip skype.com at 03:01 GMT			

**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.

Skipe manager			Avaya - Account details	- tony.skype11 - Bign out - F
🙆 🤐 😥 Feature		€73,29	Buy Skype Credit	Q, Search Nembers
B	Profile settings			B
Profile 1				
Profile settings	Profile name	Profile 1		
Authorities details	Calling character	6 channels 😔		
Automocation details	(	Next payment #25,70 to June 6, 2010		
« Back to BIP Profile list		Change subscription Cancel subscription		
	Outgoing calls	€37,54 Auto-recharge active		
	Caller ID 😡	Caller ID is set to 🐜 +16174018237		
	Incoming calls	×16174010237		
		Inhampt102		

**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.

🛆 🤮 🧭 Features	) 🖻	€73,29 Buy Skype Credit. Q. Search Members
B	Profile settings	
Profile 1	Profile name	Profile 1
Profile settings	Calling channels	6 channels 💿
Authentication details Reports	Outgoing calls	€37,54 Auto-recharge active
	(	Add credit Auto Recharge settings
+ Back to SIP Profile list		S « 28.00 Add credit Tate back credit G
	Caller ID 😡	Caller ID is set to 🔜 +10174018237
	incoming calls	<b>1</b> +16174010237
		Inhanypi102

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.

Profile 1		
	Profile name	Profile 1
Profile settings	Calling channels	6 channels 🥝
Authentication details	Outgoing calls	€37,54 Auto-recharge active
Reports	Caller ID 📀	Caller ID is set to 🖳 +16174018237
« Back to SIP Profile list	Incoming calls	+16174018237
		Inhanyp(102)
		Add a number or business account
		You can receive incoming calls on your SIP Profile via Skype Online Numbers and via Skype business accounts. When someone calls your Online Number or contacts your business account on Skype the calls get forwarded to your SIP Profile.
		Add a number Buy a new number

Figure 115 – SIP Profile for Inbound Calling

Outgoing calls	€37,54 Auto-recharge active	~
Caller ID 😨	Caller ID is set to 📟 +16174018237	~
Incoming calls		➡ +16174018237
		s nhanvpt102
		Add a number or × business account
	You can receive incoming calls on your SIP Profile via Skype Online Numbers and via S your Online Number or contacts your business account on Skype the calls get forwarde	kype business accounts. When someone calls d to your SIP Profile.
	Add Online Number	Add business account
	Add an existing business account Type an account's name Extension number (optional) ? Confirm	
	Important: If a Skype account is attached to a SIP Profile it cannot be used other device.	to sign into Skype on your computer or any

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

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

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🙆 🤐 😰 Feature	· 1		€73,29	Buy Skype Credit	Q, Search Nembers
B	Profile settings				B
Profile 1		Post d			
Des file antilene	Profile name	Profile 1			
Prome semings	Calling channels	6 channels 😒			
Authentication details	Outgoing calls	€37,54 Auto-recharge active			
Reports					
	Caller ID 💙	Caller ID is set to +16174018237			
<ul> <li>Back to SIP Profile list</li> </ul>	Incoming calls	= +16174018237 Online Number			
	<	Business account			
		Add a number or business account			

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.

🙆 🤐 🦻 Feature	· 🗐		€73,29	Buy Skype Credit	Q Search Hembers
Ð	Profile settings				
Profile 1	Profile name	Profile 1			
Profile settings	Calling channels	6 channela 😏			
Authentication details	Outgoing calls	637,54 Auto-recharge active			
Reports	Caller ID 💿	Galler ID is set to 🖼 +16174018237			
« Back to SIP Profile Rat		Change Caller ID (Usable Caller ID			
	<	Manage stored landine numbers			
	incoming calls	<b>*16174018237</b>			
		sharvpt102			
		Add a number or business account			

**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.
  - o Caller ID restriction.
  - o DTMF transmission.
  - Voicemail with MWI notification.
  - Codec negotiation.
- Performance tests:
  - Only tested with 3 calls at the same time.
  - Conference with 6 phones.
  - Call on-hold up to an hour.
  - Call duration up to 4 hours.
- Negative testing:
  - Disable IP connectivity to simulate the drop of SIP trunk.
  - Authentication challenge.
  - Account out of balance.

#### 6.2. Test Results

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

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

- <u>Issue 02.</u> 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.
- <u>Issue 06.</u> 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>;

- <u>Issue 07.</u> 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.
- <u>Issue 09.</u> Make an abandoned call from CS1000 SIP-Line phone to PSTN phone. PSTN phone rings and then CS1000 SIP-Line phone hangs up before PSTN phone answers. The call is released. However, SIP-Line phone is not able to make any other call right after that. It displays "Temporily unavailable" when making outbound calls. Wait for around 30 second, CS1000 SIP-Line phone will be back to normal status and can make outbound call.
- Issue 010. The CS1000 phone can not make any outbound calls when it is assigned to an online number to present to PSTN phones. For more detail, please refer to CR Q02164502. This issue is fixed by the patch MPLR30291 applied on CS1000 SSG.
- Issue 011. The CS1000 phone can not make any outbound calls when it is configured with caller ID restriction feature. For more detail, please refer to CR Q02164507. This issue is fixed by the patch MPLR30291 applied on CS1000 SSG.

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- 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 IP Phone 1120E Regular online 096-00-01-24 C60 47.248.101.117 1120 IP Phone 2002 Phase 2 Regular online 096-00-01-06 DCJ 47.248.101.120 2002P2 47.248.101.116 IP Phone 1140E 1140 Regular online 096-00-01-26 C60 47.248.101.115 IP Phone 1220 1220 Regular online 096-00-01-05 C6O

Verify status of digital phone registred.

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

.....

Verify status of Analog (Fax machine registered). LD 32 .stat 4 0 8 00 = UNIT 00 = IDLE (L500) 01 = UNIT 01 = IDLE (L500)

Verify the following basic calls in local CS1000.

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IP phone	call	IP phone
IP phone	call	SIP Line Client
IP Phone	call	Analog/Fax phone
IP Phone	call	Digital phone
SIP Line Client	call	Analog/Fax phone
SIP Line Client	call	Digtal Phone
Analog/Fax phone	call	Digital Phone
User can verify the same	e for calls from	n 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 RegTy	pe State	Regd-TN	UNIStimVsn
47.248.101.117	IP Phone 1120E	1120 R	egular busy	096-00-01-24	C6O
47.248.101.120	IP Phone 2002 Pha	use 2 2002P2	2 Regular b	usy 096-00-01	-06 DCJ
47.248.101.116	IP Phone 1140E	1140 R	egular busy	096-00-01-26	C6O
47.248.101.115	IP Phone 1220	1220 Re	gular busy	096-00-01-05	C6O

#### Verfify status of digital phone is busy

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

..... Verify status analog phone is busy

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

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

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

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

- 7.2. Verify that calls are established with two-way voice path when making calls from PSTN phone to Avaya phones on the CS1000 through Skype Connect Service via configured SIP trunk.
  - Verify basic call between PSTN phones and Avaya phones. At the CS1000 SIP Gateway during the call, use pcap tool (ethereal/wireshark) to make sure that all SIP request/response messages
  - Verify Codec, SIP trunk status when call is established under CS1000 call server by tracing DID number

LD 80 .trac 0 496856

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

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

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

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

LD 80 .trac 0 496856 (DID number)

• **IDLE** VTN 096 0 01 06 MARP

# 8. Conclusion

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

# 9. Additional References

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

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# **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 Q01981776-0	01 ISS1:10F1	p29065_1 27/08/2010 p29065_1.cpm NO
001 Q00349046-0	03 ISS1:10F1	p17588_1 27/08/2010 p17588_1.cpm NO
002 Q01680019	ISS1:10F1	p24307_1 27/08/2010 p24307_1.cpm NO
003 Q01725096-0	03 ISS1:10F1	p23200_1 27/08/2010 p23200_1.cpm NO
004 Q01983521-0	04 ISS1:10F1	p27616_1 27/08/2010 p27616_1.cpm NO
005 Q01849803	ISS1:10F1	p28064_1 27/08/2010 p28064_1.cpm YES
006 Q01976701-0	01 ISS1:10F1	p28211_1 27/08/2010 p28211_1.cpm NO
007 Q02024135-0	04 ISS1:10F1	p28381_1 27/08/2010 p28381_1.cpm YES
008 Q02097405	ISS1:10F1	p24463_1 27/08/2010 p24463_1.cpm NO
009 Q02029209	ISS1:10F1	p28469_1 27/08/2010 p28469_1.cpm NO
010 Q02023636	ISS1:10F1	p28475_1 27/08/2010 p28475_1.cpm NO
011 Q02022264	ISS1:10F1	p28486_1 27/08/2010 p28486_1.cpm NO
012 Q02030977	ISS1:10F1	p28507_1 27/08/2010 p28507_1.cpm NO
013 Q02020526	ISS1:10F1	p28537_1 27/08/2010 p28537_1.cpm NO
014 Q02031323-0	01 ISS1:1of1	p28546_1 27/08/2010 p28546_1.cpm NO
015 Q02034083	ISS1:10F1	p28553_1 27/08/2010 p28553_1.cpm YES
016 Q02028560-0	04 ISS1:10F1	p28564_1 27/08/2010 p28564_1.cpm NO
017 Q02034835	ISS1:10F1	p28569_1 27/08/2010 p28569_1.cpm YES

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018 Q02033951 ISS1:10F1 p2	8579_1 27/08/2010 p28579_1.cpm NO
019 Q02033139 ISS1:10F1 p2	8582_1 27/08/2010 p28582_1.cpm NO
020 Q01782930-01 ISS1:1OF1 p	24964_1 27/08/2010 p24964_1.cpm NO
021 Q02018384 ISS1:10F1 p2	8598_1 27/08/2010 p28598_1.cpm NO
022 Q02033201 ISS1:10F1 p2	8631_1 27/08/2010 p28631_1.cpm YES
023 Q02089407 ISS1:10F1 p2	9311_1 27/08/2010 p29311_1.cpm NO
024 Q02038675 ISS1:10F1 p2	8665_1 27/08/2010 p28665_1.cpm YES
025 Q02020734-02 ISS1:10F1 p	28668_1 27/08/2010 p28668_1.cpm NO
026 Q02038440 ISS1:10F1 p2	8674_1 27/08/2010 p28674_1.cpm NO
027 Q02035396 ISS1:10F1 p2	8675_1 27/08/2010 p28675_1.cpm NO
028 Q02038482 ISS1:10F1 p2	8682_1 27/08/2010 p28682_1.cpm NO
029 Q02039994 ISS1:10F1 p2	8690_1 27/08/2010 p28690_1.cpm NO
030 Q02024455-01 ISS1:10F1 p	28717_1 27/08/2010 p28717_1.cpm NO
031 Q02031359 p28679 p287	25_1 27/08/2010 p28725_1.cpm YES
032 Q02083694 ISS1:10F1 p2	9741_1 01/09/2010 p29741_1.cpm NO
034 Q02108554 ISS1:10F1 p2	9534_1 27/08/2010 p29534_1.cpm NO
036 Q01974383-02 ISS1:10F1 p	27378_1 27/08/2010 p27378_1.cpm NO
037 Q02092594 ISS1:10F1 p2	7830_1 27/08/2010 p27830_1.cpm NO
038 Q01999478-01 ISS1:1OF1 p	27897_1 27/08/2010 p27897_1.cpm NO
040 Q02007976-03 ISS1:10F1 p	28028_1 27/08/2010 p28028_1.cpm NO
041 Q02007476 ISS1:10F1 p2	8031_1 27/08/2010 p28031_1.cpm NO
042 Q02011613-01 ISS1:10F1 p	28108_1 27/08/2010 p28108_1.cpm NO
043 Q02017013-01 ISS1:10F1 p	28313_1 27/08/2010 p28313_1.cpm NO
044 Q02097631 ISS1:10F1 p2	8328_1 27/08/2010 p28328_1.cpm NO
045 Q01987270-02 ISS1:1OF1 p	28416_1 27/08/2010 p28416_1.cpm NO
046 Q01938235-05 ISS2:10F1 p	28418_2 27/08/2010 p28418_2.cpm NO
047 Q02032955-02 ISS1:10F1 p	28529_1 27/08/2010 p28529_1.cpm NO
048 Q02019323-01 ISS1:10F1 p	28551_1 27/08/2010 p28551_1.cpm NO
049 Q02100914 ISS1:10F1 p2	8597_1 27/08/2010 p28597_1.cpm NO
050 Q02032155 p28538 p286	538_1 27/08/2010 p28638_1.cpm YES
051 Q02040015 ISS1:10F1 p2	8657_1 27/08/2010 p28657_1.cpm NO
052 Q02031118 ISS1:10F1 p2	8680_1 27/08/2010 p28680_1.cpm NO
053 Q02029228-01 ISS1:1OF1 p	28681_1 27/08/2010 p28681_1.cpm NO
054 Q02043231 ISS1:10F1 p2	8712_1 27/08/2010 p28712_1.cpm NO

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055 Q02041981 p28695_1	p28719_1 27/08/2010 p28719_1.cpm NO
056 Q02031959 ISS1:10F1	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 ISS1:10F1	p28760_1 27/08/2010 p28760_1.cpm NO
059 Q02043669 ISS1:10F1	p28771_1 27/08/2010 p28771_1.cpm NO
060 Q02021470-02 ISS1:10F1	p28776_1 27/08/2010 p28776_1.cpm NO
061 Q02033321 ISS1:10F1	p28801_1 27/08/2010 p28801_1.cpm NO
062 Q02035555 ISS1:10F1	p28814_1 27/08/2010 p28814_1.cpm NO
063 Q02049121-01 ISS1:10F1	p28819_1 27/08/2010 p28819_1.cpm NO
064 Q01986974-05 ISS1:10F1	p28821_1 27/08/2010 p28821_1.cpm YES
065 Q02031502 ISS1:10F1	p28832_1 27/08/2010 p28832_1.cpm YES
066 Q02039427-02 ISS1:10F1	p28849_1 27/08/2010 p28849_1.cpm NO
067 Q02095838 1SS1:1OF1	p28852_1 27/08/2010 p28852_1.cpm NO
068 Q02036885-02 ISS1:10F1	p28857_1 27/08/2010 p28857_1.cpm NO
069 Q02043398 ISS1:10F1	p28869_1 27/08/2010 p28869_1.cpm NO
070 Q02055997 ISS1:10F1	p28895_1 27/08/2010 p28895_1.cpm NO
071 Q02044341 ISS1:10F1	p28957_1 27/08/2010 p28957_1.cpm NO
072 Q02058567-01 ISS1:10F1	p28965_1 27/08/2010 p28965_1.cpm NO
073 Q02048680 ISS1:10F1	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 ISS1:10F1	p29027_1 27/08/2010 p29027_1.cpm NO
076 Q02041385-02 ISS1:10F1	p29032_1 27/08/2010 p29032_1.cpm NO
079 Q02043226-02 ISS1:10F1	p29125_1 27/08/2010 p29125_1.cpm NO
080 Q02074796 ISS1:10F1	p29126_1 27/08/2010 p29126_1.cpm NO
081 Q02084339-02 ISS1:10F1	p29137_1 27/08/2010 p29137_1.cpm NO
082 Q02076740 ISS1:10F1	p29154_1 27/08/2010 p29154_1.cpm NO
083 Q02071451 ISS1:10F1	p29164_1 27/08/2010 p29164_1.cpm NO
084 Q02077171 ISS1:10F1	p29169_1 27/08/2010 p29169_1.cpm NO
086 Q02064503 ISS1:10F1	p29196_1 27/08/2010 p29196_1.cpm NO
087 Q02073690 ISS1:10F1	p29208_1 27/08/2010 p29208_1.cpm NO
088 Q02035822-01 ISS1:10F1	p29212_1 27/08/2010 p29212_1.cpm NO
090 Q02065521 ISS1:10F1	p29218_1 27/08/2010 p29218_1.cpm NO
091 Q02083027 ISS1:10F1	p29233_1 27/08/2010 p29233_1.cpm NO
092 Q02079849 ISS1:10F1	p29238_1 27/08/2010 p29238_1.cpm NO

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093	Q02086333	ISS1:10F1	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	ISS1:10F1	p29320_1 27/08/2010 p29320_1.cpm NO
096	Q02092223	ISS1:1of1	p29343_1 27/08/2010 p29343_1.cpm NO
097	Q02093188	ISS1:10F1	p29352_1 27/08/2010 p29352_1.cpm NO
098	Q02093256-03	ISS1:10F1	p29354_1 27/08/2010 p29354_1.cpm NO
099	Q02093325	ISS1:10F1	p29355_1 27/08/2010 p29355_1.cpm NO
100	Q02012100-06	ISS1:10F1	p29368_1 27/08/2010 p29368_1.cpm NO
101	Q02094012	ISS1:10F1	p29370_1 27/08/2010 p29370_1.cpm YES
103	Q02089914	ISS1:10F1	p29406_1 27/08/2010 p29406_1.cpm NO
104	Q02096318	ISS1:1of1	p29423_1 27/08/2010 p29423_1.cpm NO
105	Q02097948	ISS1:10F1	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	ISS1:10F1	p29464_1 27/08/2010 p29464_1.cpm NO
109	Q02103928	ISS1:10F1	p29486_1 27/08/2010 p29486_1.cpm NO
111	Q02104745-01	ISS1:10F1	p29495_1 27/08/2010 p29495_1.cpm NO
115	Q02109161	ISS1:10F1	p29536_1 27/08/2010 p29536_1.cpm NO
117	Q02119261	ISS2:10F1	p29613_2 27/08/2010 p29613_2.cpm NO
122	Q02096730	p29462 p2855	57 p29676_1 27/08/2010 p29676_1.cpm NO
124	Q02024749-02	ISS1:10F1	p29680_1 27/08/2010 p29680_1.cpm NO
125	Q02110973	ISS1:10F1	p29690_1 27/08/2010 p29690_1.cpm NO
128	Q02096711	ISS1:10F1	p29714_1 27/08/2010 p29714_1.cpm NO
129	Q02114752	ISS1:10F1	p29718_1 27/08/2010 p29718_1.cpm NO
130	Q02122052	ISS1:10F1	p29726_1 27/08/2010 p29726_1.cpm NO
131	Q02124953	ISS1:10F1	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	ISS1:10F1	p29802_1 27/08/2010 p29802_1.cpm NO
134	Q02108852	ISS1:10F1	p29825_1 27/08/2010 p29825_1.cpm NO
135	Q02129264	ISS1:10F1	p29827_1 27/08/2010 p29827_1.cpm NO
136	Q02128131	ISS1:10F1	p29830_1 27/08/2010 p29830_1.cpm NO
137	Q02111317	ISS1:10F1	p29844_1 27/08/2010 p29844_1.cpm NO
138	Q02131547	ISS1:10F1	p29880_1 27/08/2010 p29880_1.cpm NO
138 139	Q02131547 Q02135191	ISS1:10F1 ISS1:10F1	p29880_1 27/08/2010 p29880_1.cpm NO p29935_1 27/08/2010 p29935_1.cpm NO

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 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 p28797 1 Yes 20 FRU nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386 10/08/10 NO p29703 1 Yes 21 10/08/10 NO FRU nortel-cs1000-shared-ssSubagent-6.00.18-00.i386 22 p25946\_1 Yes 10/08/10 NO FRU nortel-cs1000-pi-control-1.00.00.00-00.noarch nortel-cs1000-pi-control-1.00.00.00-00.noarch 24 p27159\_1 Yes 10/08/10 NO FRU 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 10/08/10 yes 0 Yes nortel-cs1000-linuxbase-6.00.18.65-03.i386.001 yes 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 YES nortel-cs1000-sps-6.00.18.63-00.i386.000 10/08/10 NO 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 nortel-cs1000-dmWeb-6.00.18.62-00.i386.001 Yes 10/08/10 NO 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 Yes YES 12 10/08/10 NO nortel-cs1000-dbcom-6.00.18.65-01.i386.001 13 YES Yes 10/08/10 YES nortel-cs1000-tps-6.00.18.65-07.i386.000 14 Yes 10/08/10 YES YES nortel-cs1000-csv-6.00.18.65-04.i386.000 16 Yes 10/08/10 NO YES nortel-cs1000-bcc 6-0-6.00.18.65-02.i386.000 17 Yes 10/08/10 NO YES nortel-cs1000-cs1000WebService 6-0-6.00.18.65-02.i386.000 18 Yes 10/08/10 NO YES nortel-cs1000-ftrpkg-6.00.18.65-02.i386.000 25 Yes 10/08/10 NO YES nortel-cs1000-vtrk-6.00.18.65-TMP297.i386.001

[nortel@node1-carrier ~]\$

#### **Appendix C: Configure SIP trunk in CS1000 using overlays**

*Procedure summary* 

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

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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
СТҮР	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).

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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
ТКТР	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.

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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.
СНТҮ	ВСН	B-channel type.
СТҮР	CDP	Coordinated dialing plan

### **Define virtual trunks**

#### Use Overlay 14

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

# Appendix D: Sample inbound and outbound SIP Invite

# A sample Inbound SIP Invite

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

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

Status-Line: SIP/2.0 100 Trying From: <sip:16139675281@sip.skype.com>;tag=a4a109cc-13c4-4c73ca7f-1b7008e0-415501ca To: <sip:13157914465@sip.skype.com:5060> Call-ID: CXC-436-68081050-a4a109cc-13c4-4c73ca7f-1b7008e0-4082af6c CSeq: 1 INVITE Via: SIP/2.0/UDP 204.9.161.164:5060;branch=z9hG4bK-2852c-4c73ca7f-1b7008e0-63f97f3b Supported: 100rel,x-nortel-sipvc,replaces,timer User-Agent: Nortel CS1000 SIP GW release\_6.0 version\_ssLinux\_6.00.18

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

Status-Line: SIP/2.0 200 OK

### A sample outbound SIP Invite

Request-Line: INVITE sip:16139675281@sip.skype.com;user=phone SIP/2.0 From: "Skype 1120"<sip:99051000106920@sip.skype.com>;tag=b0fbe360-f464f82f-13c4-40030-a70fe-2a120f2c-a70fe To: <sip:16139675281@sip.skype.com;user=phone> Call-ID: b0ca0718-f464f82f-13c4-40030-a70fe-5ac2cda6-a70fe CSeq: 1 INVITE Via: SIP/2.0/UDP 47.248.100.244:5060;branch=z9hG4bK-a70fe-28c962d8-141e4445 Max-Forwards: 70 Supported: 100rel,x-nortel-sipvc,replaces,timer User-Agent: Nortel CS1000 SIP GW release 6.0 version ssLinux 6.00.18 P-Asserted-Identity: "Skype 1120"<sip:13157914465@sip.skype.com;user=phone> Privacy: none x-nt-e164-clid: +113157914465@sip.skype.com;user=phone History-Info: <sip:16139675281@sip.skype.com;user=phone>;index=1 Alert-Info: cid:external@sip.skype.com x-nt-corr-id: 000002880918131808@0019e1e82491-eccec122 Contact: <sip:13157914465@sip.skype.com:5060;maddr=47.248.100.244;transport=udp;user=phone>

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

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

Status-Line: SIP/2.0 200 OK

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