



>BUSINESS MADE **SIMPLE**

**NORTEL**

Nortel Communication Server 1000 Release 6.0 Interoperability Config Guide

Cisco Unified Communications Manager Release 6.0.1 via SIP

July 15, 2009

Issue: 1.0

Document Number NN70003-001



## **Copyright © 2008-2009 Nortel Networks**

All rights reserved. July 2009

The information in this document is subject to change without notice. The statements, configurations, technical data, and recommendations in this document are believed to be accurate and reliable, but are presented without express or implied warranty. Users must take full responsibility for their applications of any products specified in this document. The information in this document is proprietary to Nortel Networks Inc.

The software described in this document is furnished under a license agreement and may be used only in accordance with the terms of that license.

## **Trademarks**

Nortel, the Nortel logo, the Globemark, Unified Networks, and BayStack are trademarks of Nortel Networks.

Cisco, Cisco Systems, Communications Manager, Unified Communications Manager are trademarks of Cisco Systems.

Adobe and Acrobat Reader are trademarks of Adobe Systems Incorporate.

All other Trademarks are the property of their respective owners.



# Table of Contents

<u>INTRODUCTION</u> .....	3
<u>BASIC CONFIGURATION NOTES</u> .....	4
<u>HARDWARE AND SOFTWARE VERSIONS</u> .....	4
<u>FEATURES OVER SIP TRUNK</u> .....	5
<u>INTEGRATION NOTES AND LIMITATIONS</u> .....	6
<u>NETWORK TOPOLOGY</u> .....	8
<u>NORTEL COMMUNICATION SERVER 1000 SETUP</u> .....	9
<u>PHONE CONFIGURATION ON CISCO UNIFIED COMMUNICATIONS MANAGER</u> .....	10
<u>DNS AND DOMAIN CONFIGURATION ON CISCO UNIFIED COMMUNICATIONS MANAGER</u> .....	15
<u>DNS NAMES AND DOMAIN CONFIGURATION ON MICROSOFT WINDOWS DNS SERVER</u> .....	16
<u>MTP CONFIGURATION ON CISCO UNIFIED COMMUNICATIONS MANAGER</u> .....	17
<u>REGION CONFIGURATION ON CISCO UNIFIED COMMUNICATIONS MANAGER</u> .....	20
<u>DEVICE POOL CONFIGURATION ON CISCO UNIFIED COMMUNICATIONS MANAGER</u> .....	21
<u>SIP TRUNK PROFILE SETUP ON CISCO UNIFIED COMMUNICATIONS MANAGER</u> .....	22
<u>SIP TRUNK CONFIGURATION ON CISCO UNIFIED COMMUNICATIONS MANAGER</u> .....	23
<u>CALL ROUTING CONFIGURATION ON CISCO UNIFIED COMMUNICATIONS MANAGER</u> .....	26
<u>CLUSTER WIDE DOMAIN NAME SETUP ON CISCO UNIFIED COMMUNICATIONS MANAGER</u> .....	28
<u>ZONE CONFIGURATION ON NORTEL COMMUNICATION SERVER 1000</u> .....	29
<u>NODE IP CONFIGURATION ON NORTEL COMMUNICATION SERVER 1000</u> .....	30
<u>VGW AND IP PHONE CODEC CONFIGURATION ON NORTEL COMMUNICATION SERVER 1000</u> .....	32
<u>QOS CONFIGURATION ON NORTEL COMMUNICATION SERVER 1000</u> .....	33
<u>LAN CONFIGURATION ON NORTEL COMMUNICATION SERVER 1000</u> .....	34
<u>SIP GW CONFIGURATION ON NORTEL COMMUNICATION SERVER 1000</u> .....	35
<u>SIGNALLING SERVER CONFIGURATION ON NORTEL COMMUNICATION SERVER 1000</u> .....	36
<u>(MGC) MEDIA GATEWAY CONTROLLER CONFIGURATION ON NORTEL COMMUNICATION SERVER 1000</u> .....	37
<u>(SPS) SIP PROXY SERVER SYSTEM WIDE CONFIGURATION ON COMMUNICATION SERVER 1000</u> .....	39
<u>(SPS) SIP PROXY SERVER CONFIGURATION ON COMMUNICATION SERVER 1000</u> .....	40
<u>SERVICE DOMAIN CONFIGURATION ON NORTEL COMMUNICATION SERVER 1000</u> .....	42
<u>L1 (UDP) DOMAIN CONFIGURATION ON NORTEL COMMUNICATION SERVER 1000</u> .....	42
<u>L0 (CDP) DOMAIN CONFIGURATION ON NORTEL COMMUNICATION SERVER 1000</u> .....	43
<u>SIP GATEWAY ENDPOINT CONFIGURATION ON NORTEL COMMUNICATION SERVER 1000</u> .....	44
<u>SIP ROUTING ENTERIES CONFIGURATION ON NORTEL COMMUNICATION SERVER 1000</u> .....	45
<u>IP D-CHANNEL CONFIGURATION ON NORTEL COMMUNICATION SERVER 1000</u> .....	46
<u>SUPER-LOOP CONFIGURATION ON NORTEL COMMUNICATION SERVER 1000</u> .....	49
<u>SIP ROUTES AND TRUNKS CONFIGURATION ON NORTEL COMMUNICATION SERVER 1000</u> .....	51
<u>MGC TRUNK CONFIGURATION ON NORTEL COMMUNICATION SERVER 1000</u> .....	59
<u>CDP STEERING CODE CONFIGURATION ON NORTEL COMMUNICATION SERVER 1000</u> .....	63



## **Introduction**

The purpose of this document is to detail the steps and configuration necessary for Nortel Communication Server 1000 Release 6.0 to interoperate with Cisco Unified Communications Manager running software release 6.0.1 over SIP Trunks only. There is a known issue to interop with Cisco Unified Communications Manager over H323 trunk as Cisco uses different ports in OLC and OLCAck. That cause one way speech issue for any call involving hold and retrieve over H323. **So we recommend using SIP or QSIG to interop with Cisco Unified Communications Manager until the issue is fixed.**

This document doesn't include every possible combination of hardware, software, Protocol or feature testing scenarios.

## **Basic Configuration Notes**

This application note is based on the configuration shown in the diagram on page 9.

Default configurations are used on both systems with the exception of the following:

- Cisco Unified Communications Manager is configured as a Static SIP end point to Nortel SPS.
- Cisco Unified Communications Manager is also configured as (MTP) media Termination Point. (Required)
- G711 codec with 20ms payload on CCM, 30 and 10 ms also work.
- CDP dialing plan on Nortel

## **Hardware and Software Versions**

### **Nortel**

<b><u>Quantity</u></b>	<b><u>Hardware</u></b>	<b><u>Software Version</u></b>
1	Nortel (CS 1000) Communication server 1000 (CPPM)	6.0
1	Signaling Servers (CPPM)	6.0
1	(SPS) SIP proxy Server HP DL320	6.0
1	CallPilot for messaging (TRP) 703	05.00.41.30
2	1140 IP sets	
1	T7316 Nortel Digital Phone	
1	Nortel Call Server Patch(s)	Deplist
3	Signaling Server Patch(s)	Deplist

### **Cisco**

<b><u>Quantity</u></b>	<b><u>Hardware</u></b>	<b><u>Software Version</u></b>
1	Cisco Unified Communications Manager MCS7825H- 3.0-IPC1, DL320 G2	6.1.3.3190-1
2	7960 IP sets (SCCP)	cmterm-7940-7960-sccp.8-0-7.cop
1	7960 IP set (SIP)	cmterm-7940-7960-8.8.00-sip.cop
1	CCM patch	N/A

### **LAN Infrastructure**

<b><u>Quantity</u></b>	<b><u>Hardware</u></b>	<b><u>Software Version</u></b>
1	Cisco 6503 Router	(C6MSFC2-PSV-M), Version 12.1(19)E1
1	Cisco 4503 L2 Switch	8.4(2)GLX
1	Nortel Baystack 5520 L2 Switch	5.0.5.020
1	DNS Server (Dell PC)	Windows 2000



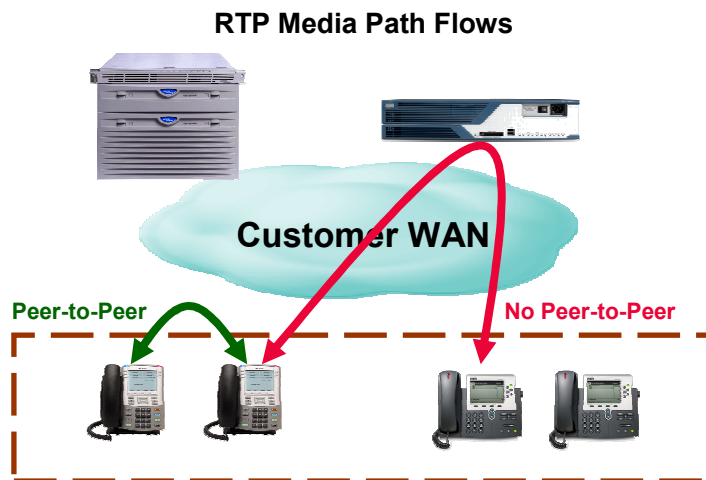
## **Features Over SIP Trunk**

- Basic Call
- Hold and Retrieve
- Attended Transfer
- Blind Transfer
- Call Forward
- Conference
- Personal Call Assistant (PCA)
- Caller Line ID (CLID) and Name Display
- Messaging CallPilot
- RFC2833
- G711 Codec with 30ms, 20ms and 10ms payload.
- Secured Reliable Transfer Protocol (SRTP)

## **Integration Notes and Limitations**

### **1) No direct media path for IP Trunking**

The Nortel Communication server 1000 supports a direct peer-to-peer media path between all IP endpoints. When these two systems are interconnected, the Cisco Unified Communications Manager requires the media path to pass through the Cisco Unified Communications Manager. This may have a significant impact on WAN bandwidth engineering if the voice packets need to be backhauled to Cisco Unified Communications Manager in a remote location as in the diagram below.



### **2) Cisco Media Termination Points and G.711 requirement**

To interoperate with the Nortel Communication server 1000 the Cisco Unified Communications Manager requires Media Termination Points (MTPs) to provide any supplementary features beyond basic call setup and teardown. While the Nortel Communication server 1000 is able to support and intelligently negotiate different codecs, the Cisco Unified Communications Manager's software based MTP will only support G.711. Special consideration may need to be given for additional bandwidth requirements.

### **3) Calling Party Name Display and Caller ID**

The Nortel Communication server 1000 uses the standard SIP "**P-Asserted-Id**" field to pass Calling Party Name Display as well as Caller ID information across SIP trunk resources to other nodes in the network. The Cisco Unified Communications Manager employs a different implementation, and therefore is unable to pass along this information to the Communication server 1000. This will have a negative impact to call routing for Caller ID sensitive applications such as a Contact Center and proper E911 call recognition and termination.

### **4) Call Redirect Information**

Call redirect information is used by various other applications such as CallPilot to terminate the call directly into the called party's mailbox. Interworking between these two devices limits critical information over SIP trunk, preventing calls from properly terminating.



#### **5) SRTP on IP Trunks**

The Cisco Unified Communications Manager does not currently support the SRTP protocol over IP trunks. Calls originated from clients on the Cisco Unified Communications Manager with SRTP are passed over IP Trunks in an unencrypted state. The Nortel clients must be set for “Best Effort” security for SRTP so that proper negotiation can take place between the Cisco and Nortel clients.

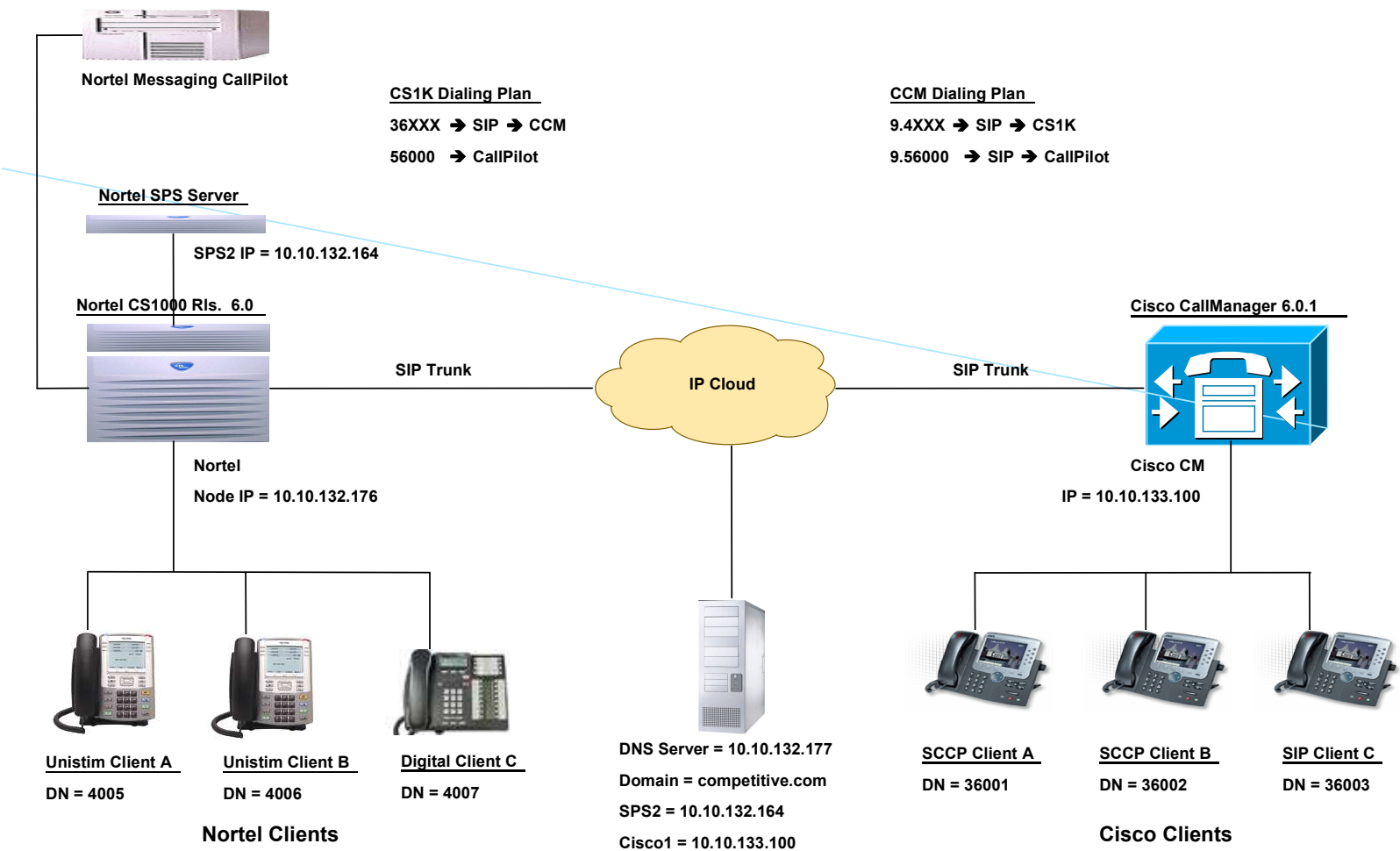
#### **6) DTMF over SIP to Callpilot**

In Cisco Callmanager 5.0, DTMF from Cisco Client can't be passed to Nortel Callpilot properly. However, this is no longer an issue with Cisco Unified Communications Manager 6.0.



## Network Topology

### Nortel-Cisco End-to-End Call Setup Configuration using SPS and SIP Trunk





## **CISCO UNIFIED COMMUNICATIONS MANAGER SETUP**

- 1) Create Cisco IP Clients.
- 2) Configure Cisco IP phone and line DN
- 3) Configure Cisco Unified Communications Manager DNS and Domain settings
- 4) Configure Microsoft Windows 2000 DNS Server settings
- 4) Create the Media Resource Group and Media Resource Group List for the MTP requirements.
- 5) Create Region.
- 6) Create Device Pool.
- 7) Create SIP Trunk Profile.
- 8) Add a SIP Trunk for Nortel Communication server 1000 under the Device pull-down menu.
- 9) Add a Route Pattern to reach the Nortel's phone DN extensions.
- 10) Add a Route Pattern to reach the Nortel CallPilot VoiceMail system.
- 11) Create Top level (SIP) Domain in Enterprise Parameters.

## **Nortel Communication server 1000 Setup**

### **Signaling Server Setup Sequence using Nortel Element Manager**

- 1) Configure the Zones
- 2) Configure a new IP Telephony Node summary
- 3) Configure the Node section
- 4) Configure the VGW and IP phone codec profile section
- 5) Configure the Quality of Service (QoS) section. (Use default)
- 6) Configure LAN Configuration section
- 7) Configure the SIP GW Setting section (Top level Domain must match CCM)
- 8) Configure the Signaling Server section
- 9) Configure (MGC) Media Gateway Controller (Required for Conference Calls)

### **(SPS) SIP Proxy Server Setup Sequence using Web Browser**

- 1) Configure the System Wide Settings
- 2) Configure the SPS Server Settings
- 3) Configure a Service Domain
- 4) Configure a L1 Domain (UDP)
- 5) Configure a L0 Domain (CDP)
- 6) Configure a SIP Gateway Endpoint
- 7) Configure the Routing Entries for SIP

### **Call Server Setup Sequence using CPPM Card Console**

- 1) Configure the IP D-channel (signaling channel) between the Call Server and the Signaling Server
- 2) Configure the Super-loop for the Virtual Trunks
- 3) Configure the SIP Virtual Trunks to the Signaling Server
- 4) Configure the Virtual Gateway Trunks
- 5) Configure the Virtual lines for the Nortel IP phone
- 6) Configure the SIP route
- 7) Configure the Route List Block for the Virtual Trunk route

## 8) Configure CDP steering codes

Cisco Unified Communications Manager Configuration Sequence.**Phone Configuration on Cisco Unified Communications Manager**






Use the Cisco Unified Communications Manager Administration Phone Configuration window to configure Cisco Unified IP Phones and devices:

From Cisco Unified Communications Manager Administration Window drop down menu. Select:- Device / Phone / Add New, Select the phone type from drop down menu and the protocol, SIP or SCCP.

Association Information		Phone Type	
<div>Modify Button Items</div> <div> <div>1</div> <div>7710 7715</div> <div>Line [1] - 47002 (no partition)</div> </div> <div> <div>2</div> <div>7710 7715</div> <div>Line [2] - 36002 (no partition)</div> </div> <div> <div>3</div> <div>7710 7715</div> <div>Line [3] - 47050 (no partition)</div> </div> <div> <div>4</div> <div>7710 7715</div> <div>Line [4] - 36099 (no partition)</div> </div> <div> <div>5</div> <div>7710 7715</div> <div>Add a new SD</div> </div> <div> <div>6</div> <div>7710 7715</div> <div>Add a new SD</div> </div> <div> <div>----- Add On Module(s) -----</div> </div> <div> <div>7</div> <div>None</div> </div> <div> <div>8</div> <div>None</div> </div> <div> <div>9</div> <div>None</div> </div> <div> <div>10</div> <div>None</div> </div> <div> <div>11</div> <div>None</div> </div> <div> <div>12</div> <div>None</div> </div> <div> <div>13</div> <div>None</div> </div> <div> <div>14</div> <div>None</div> </div> <div> <div>15</div> <div>None</div> </div> <div> <div>16</div> <div>None</div> </div> <div> <div>17</div> <div>None</div> </div> <div> <div>18</div> <div>None</div> </div> <div> <div>19</div> <div>None</div> </div> <div> <div>20</div> <div>None</div> </div> <div> <div>21</div> <div>None</div> </div> <div> <div>22</div> <div>None</div> </div> <div> <div>23</div> <div>None</div> </div> <div> <div>24</div> <div>None</div> </div> <div> <div>25</div> <div>None</div> </div>			

Note: This is a proprietary screenshot from Cisco Unified Communications Manager 6.0

Phone Configuration Continued.

25	None
26	None
27	None
28	None
29	None
30	None
31	None
32	None
33	None
34	None
----- Unassigned Associated Items -----	
35	 <a href="#">Line [5] - Add a new DN</a>
36	 <a href="#">Add a new SD</a>
37	 <a href="#">Add a new SURL</a>
38	 <a href="#">Add a new BLF SD</a>
39	 <a href="#">Add a new BLF Directed Call Park</a>
40	Privacy
41	None

☒ Retry Video Call as Audio  
☐ Ignore Presentation Indicators (internal calls only)  
☒ Allow Control of Device from CTI  
☒ Logged Into Hunt Group  
☐ Remote Device

---

**Protocol Specific Information**

Packet Capture Mode\* None  
 Packet Capture Duration 0  
 Presence Group\* Standard Presence group  
 Device Security Profile\* Cisco 7960 - Standard SCCP Non-Secure Profile  
 SUBSCRIBE Calling Search Space < None >  
☐ Unattended Port  
☒ Require DTMF Reception  
☐ RFC2833 Disabled

---

**Certification Authority Proxy Function (CAPF) Information**

Certificate Operation\* No Pending Operation  
 Authentication Mode\* By Null String  
 Authentication String 6424334015  
  
 Key Size (Bits)\* 1024  
 Operation Completes By 2006 12 11 12 (YYYY:MM:DD:HH)  
 Certificate Operation Status: None  
 Note: Security Profile Contains Addition CAPF Settings.

---

**Expansion Module Information**

Module 1 < None >

Note: This is a proprietary screenshot from Cisco Unified Communications Manager 6.0

Phone DN Configuration.

Directory Number Information			
Directory Number*	<input type="text" value="36002"/>		
Route Partition	<input style="border: none;" type="text" value=" &lt; None &gt; "/>		
Description	<input type="text"/>		
Alerting Name	<input type="text" value="Lucy VanPelt"/>		
ASCII Alerting Name	<input type="text" value="Lucy VanPelt"/>		
<input checked="" type="checkbox"/> Allow Control of Device from CTI			
Associated Devices	<input type="text" value="SEP000FF72649C5"/>	<input type="button" value="Edit Device"/>	<input type="button" value="Edit Line Appearance"/>
	▼ ▲		
Dissociate Devices	<input type="text"/>		

Directory Number Settings	
Voice Mail Profile	<input type="text" value="VM_QSIG_CP"/> (Choose <None> to use system default)
Calling Search Space	<input style="border: none;" type="text" value=" &lt; None &gt; "/>
Presence Group*	<input type="text" value="Standard Presence group"/>
User Hold MOH Audio Source	<input style="border: none;" type="text" value=" &lt; None &gt; "/>
Network Hold MOH Audio Source	<input style="border: none;" type="text" value=" &lt; None &gt; "/>
Auto Answer*	<input type="text" value="Auto Answer Off"/>

AAR Settings			
	Voice Mail	AAR Destination Mask	AAR Group
AAR	<input type="checkbox"/> or	<input type="text"/>	<input style="border: none;" type="text" value=" &lt; None &gt; "/>
<input checked="" type="checkbox"/> Retain this destination in the call forwarding history			

**Note:** This is a proprietary screenshot from Cisco Unified Communications Manager 6.0

Phone DN Configuration Continued.**Call Forward and Call Pickup Settings**

	Voice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy			Use System Default
Forward All	<input type="checkbox"/> or		< None >
Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	<input type="checkbox"/> or		< None >
Forward Busy External	<input type="checkbox"/> or		< None >
Forward No Answer Internal	<input checked="" type="checkbox"/> or		< None >
Forward No Answer External	<input checked="" type="checkbox"/> or		< None >
Forward No Coverage Internal	<input type="checkbox"/> or		< None >
Forward No Coverage External	<input type="checkbox"/> or		< None >
Forward on CTI Failure	<input type="checkbox"/> or		< None >
Forward Unregistered Internal	<input checked="" type="checkbox"/> or		< None >
Forward Unregistered External	<input checked="" type="checkbox"/> or		< None >
No Answer Ring Duration (seconds)			
Call Pickup Group			< None >

**MLPP Alternate Party Settings**

Target (Destination)	
MLPP Calling Search Space	< None >
MLPP No Answer Ring Duration (seconds)	

**Line Settings for All Devices**

Hold Reversion Ring Duration (seconds)		Setting the Hold Reversion Ring Duration to zero will disable the feature
Hold Reversion Notification Interval (seconds)		Setting the Hold Reversion Notification Interval to zero will disable the feature

**Note:** This is a proprietary screenshot from Cisco Unified Communications Manager 6.0

## Phone DN Configuration Continued.

### Line 2 on Device SEP000FF72649C5

Display (Internal Caller ID)	<input type="text"/>	Display text for a line appearance is intended for displaying text such as a name instead of a calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID)	<input type="text"/>	
Line Text Label	<input type="text"/>	
ASCII Line Text Label	<input type="text"/>	
External Phone Number Mask	<input type="text"/>	
Visual Message Waiting Indicator Policy*	<input type="text" value="Use System Policy"/>	
Ring Setting (Phone Idle)*	<input type="text" value="Use System Default"/>	
Ring Setting (Phone Active)	<input type="text" value="Use System Default"/>	Applies to this line when any line on the phone has a call in progress.
Call Pickup Group Audio Alert Setting(Phone Idle)	<input type="text" value="Use System Default"/>	
Call Pickup Group Audio Alert Setting(Phone Active)	<input type="text" value="Use System Default"/>	
Monitoring Calling Search Space	<input type="text" value="&lt; None &gt;"/>	

### Multiple Call/Call Waiting Settings on Device SEP000FF72649C5

Note: The range to select the Max Number of calls is: 1-188

Maximum Number of Calls*	<input type="text" value="4"/>	
Busy Trigger*	<input type="text" value="2"/>	(Less than or equal to Max. Calls)

### Forwarded Call Information Display on Device SEP000FF72649C5

- ☒ Caller Name
- ☒ Caller Number
- ☒ Redirected Number
- ☒ Dialed Number

### Users Associated with Line

Associate End Users

Note: This is a proprietary screenshot from Cisco Unified Communications Manager 6.0

### **DNS and Domain Configuration on Cisco Unified Communications Manager**


Use putty or secured shell client (port 22) to login into Cisco Unified Communications Manager with administrative privileges. At the CLI command prompt use the following command to configure DNS and Domain name in which to place Communications Manager.

Note: After the changes are made reboot is required.

At the CLI command prompt type the following.

- 1) set network dns primary 10.10.132.177
- 2) set network domain competitive.com \*

Following is a GUI output to confirm changes after the reboot.

**Status**  
 Status: Ready

**Ethernet Details**  
DHCP disabled  
Status  
IP Address 10.10.133.100  
IP Mask 255.255.255.224  
Link Detected yes  
Mode  
Queue Length 1000  
MTU 1500  
MAC Address 00:11:0a:30:86:3d  
RX Stats  
TX Stats

**DNS Details**  
Primary 10.10.132.177  
Secondary  
Options timeout:5 attempts:2  
Domain competitive.com  
Gateway 47.11.133.97

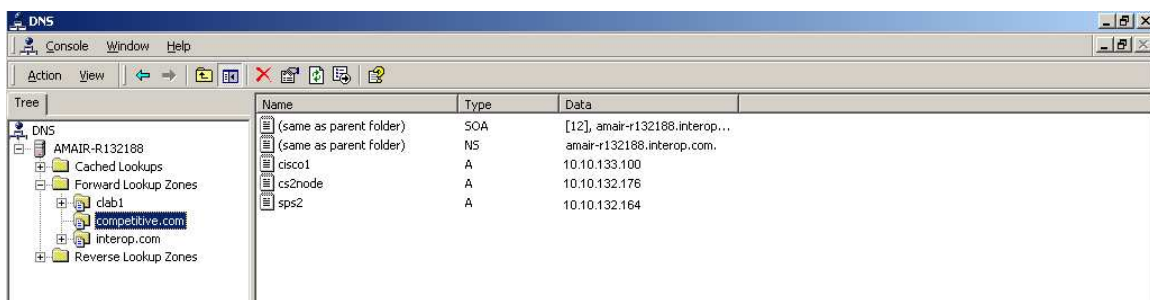
Note: This is a proprietary screenshot from Cisco Unified Communications Manager 6.0



## **DNS Names and Domain Configuration on Microsoft Windows DNS Server**

Domain = competitive.com \*

sps2.competitive.com = 10.10.132.164 \*



Note: This is a proprietary screenshot from Microsoft Windows 2000 DNS Server

\*: In this guide , we are using xyz.com as the network domain, but simply using com as the network domain will work fine as well. See example below.

Domain = com

competitive.com = 10.10.132.164

### **MTP Configuration on Cisco Unified Communications Manager**

For Nortel Communication server 1000 and Cisco Unified Communications Manager to interoperate with each other (Basic Call and/or Supplementary features), it required MTP resources.

Cisco Unified Communications Manager requires an RFC 2833 DTMF-compliant MTP device to make SIP calls. The current standard for SIP uses inband payload types to indicate DTMF tones, and Cisco IP telephony components such as SCCP IP phones support only out-of-band payload types. Thus, an RFC 2833-compliant MTP device monitors for payload type and acts as a translator between inband and out-of-band payload types.


Users should refer to Cisco Unified Communications Manager System Guide Chapter 61 for more details

[http://www.cisco.com/en/US/docs/voice\\_ip\\_comm/cucm/admin/6\\_1\\_1/ccmcfg/bccm.pdf](http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/6_1_1/ccmcfg/bccm.pdf)

#### **Create (MTP) Media Termination Point**

From Cisco Unified Communications Manager Administration Window drop down menu. Select:- Media Resources / Media Termination Point / Add New

<b>Media Termination Point Information</b>	
Registration	Registered with Cisco Unified Communications Manager 10.10.133.100
IP Address	10.10.133.100
Media Termination Point Type*	Cisco Media Termination Point Software
Host Server*	10.10.133.100
Media Termination Point Name*	<input type="text" value="MTP_CiscoCM"/>
Description	<input type="text" value="MTP_CiscoCM"/>
Device Pool*	<input type="text" value="G711_Pool"/>



 \*- indicates required item.

Note: This is a proprietary screenshot from Cisco Unified Communications Manager 6.0

## MTP Configuration Continued.


### Create Media Resource Group

Select:- Media Resources / Media Resource Group / Add New

<b>Media Resource Group Status</b>	
Media Resource Group: sw_mrg (used by 53 devices)	
<b>Media Resource Group Information</b>	
Name*	sw_mrg
Description	sw_mrg
<b>Devices for this Group</b>	
Available Media Resources**	10.10.133.100 ANN_CISCO1
▼ ▲	
Selected Media Resources*	CM_CISCO1 (CFB) MTP_CiscoCM (MTP) ▼ ▲
<input type="checkbox"/> Use Multicast for MOH Audio (If at least one multicast MOH resource is available)	
<div>Save Delete Copy Reset Add New</div>	
<p> *- indicates required item.</p> <p> **Includes Annunciators (ANN), Conference Bridges (CFB), Media Termination Points (MTP), Music On Hold Servers (MOH) and Transcoders (XCODE)</p>	

### Create Media Resource Group List

Select:- Media Resources / Media Resource Group List / Add New

<b>Media Resource Group List Status</b>	
Media Resource Group List: sw_mrg1 (used by 53 devices)	
<b>Media Resource Group List Information</b>	
Name*	<input type="text" value="sw_mrg1"/>
<b>Media Resource Groups for this List</b>	
Available Media Resource Groups	<div></div>
▼ ▲	
Selected Media Resource Groups	<div>sw_mrg</div>
▼ ▲	
<div>Save Delete Copy Reset Add New</div>	
 *- indicates required item.	

Note: These are proprietary screenshots from Cisco Unified Communications Manager 6.0

## **Region Configuration on Cisco Unified Communications Manager**

Regions are used to specify the bandwidth that is used for audio and video calls within a region and between existing regions.

From Cisco Unified Communications Manager Administration Window drop down menu.  
Select:- System / Region / Add New

**Region Information**

Name \*

**Region Relationships**

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Default	G.711	384	Use System Default
G711	G.711	384	Use System Default
G723	G.723	Use System Default	Use System Default
G729	G.729	Use System Default	Use System Default

NOTE: Regions(s) not displayed
Use System Default
Use System Default
Use System Default

**Modify Relationship to other Regions**

Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
<input type="text" value="Default"/> <input type="text" value="G711"/> <input type="text" value="G723"/> <input type="text" value="G729"/>	<input type="text" value="Keep Current Setting"/>	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text" value=""/> kbps	<input type="text" value="Keep Current Setting"/>

*i* \* - indicates required item.

*i* \*\*The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 Kbps between regions and can be used interchangeably.

Note: This is a proprietary screenshot from Cisco Unified Communications Manager 6.0



### **Device Pool Configuration on Cisco Unified Communications Manager**

Use device pools to define sets of common characteristics for devices. You can specify the following device characteristics for a device pool:

- Cisco Unified Communications Manager Group
- Date/time group
- Region
- Softkey template
- SRST reference
- Calling search space for auto-registration
- Media resource group list
- Music On Hold (MOH) audio sources
- User and network locales
- Connection monitor duration timer for communication between SRST and Cisco Unified Communications Manager
- MLPP settings

From Cisco Unified Communications Manager Administration Window drop down menu. Select:- System / Device Pool / Add New

Device Pool Settings	
Device Pool Name*	G711_Pool
Cisco Unified Communications Manager Group*	Default
Calling Search Space for Auto-registration	< None >
Reverted Call Focus Priority	Default

Roaming Sensitive Settings	
Date/Time Group*	CMLocal
Region*	G711
Media Resource Group List	sw_mrg1
Location	< None >
Network Locale	United States
SRST Reference*	Disable
Connection Monitor Duration***	120
Physical Location	< None >
Device Mobility Group	< None >

Device Mobility Related Information****	
Device Mobility Calling Search Space	< None >
AAR Calling Search Space	< None >
AAR Group	< None >


Note: This is a proprietary screenshot from Cisco Unified Communications Manager 6.0

### **SIP Trunk Profile Setup on Cisco Unified Communications Manager**

The SIP Trunk Security Profile window includes security-related settings such as transport type, device security mode, digest authentication settings, and authorization settings for incoming SIP messages. You must apply a security profile to all SIP trunks that are configured in Cisco Unified Communications Manager Administration.

From Cisco Unified Communications Manager Administration Window drop down menu. Select:- System / Security Profile / SIP Trunk Security Profile / Add New

#### **For a Non Secure SIP Trunk**

**Status**
  
 Status: Ready

---

**SIP Trunk Security Profile Information**
  

Name*	<input type="text" value="SIP_Security_Profile_I"/>
Description	<input type="text" value="Non Secure_Port_5060"/>
Device Security Mode	<input type="text" value="Non Secure"/>
Incoming Transport Type*	<input type="text" value="TCP+UDP"/>
Outgoing Transport Type	<input type="text" value="TCP"/>
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	<input type="text" value="600"/>
X.509 Subject Name	<input type="text"/>
Incoming Port*	<input type="text" value="5060"/>
<input type="checkbox"/> Enable Application Level Authorization	
<input type="checkbox"/> Accept Presence Subscription	
<input type="checkbox"/> Accept Out-of-Dialog REFER	
<input type="checkbox"/> Accept Unsolicited Notification	
<input type="checkbox"/> Accept Replaces Header	

Note: This is a proprietary screenshot from Cisco Unified Communications Manager 6.0

### **SIP Trunk Configuration on Cisco Unified Communications Manager**

Use a trunk device to configure a logical route to a gatekeeper (that is, the wholesale network or an intercluster trunk with gatekeeper control), to an intercluster trunk without a gatekeeper, or to a SIP network. For trunk to Nortel CS 1000 use FQDN of SPS.

From Cisco Unified Communications Manager Administration Window drop down menu. Select:- Device / Trunk / Add New / Trunk Type = SIP / Device Protocol = SIP





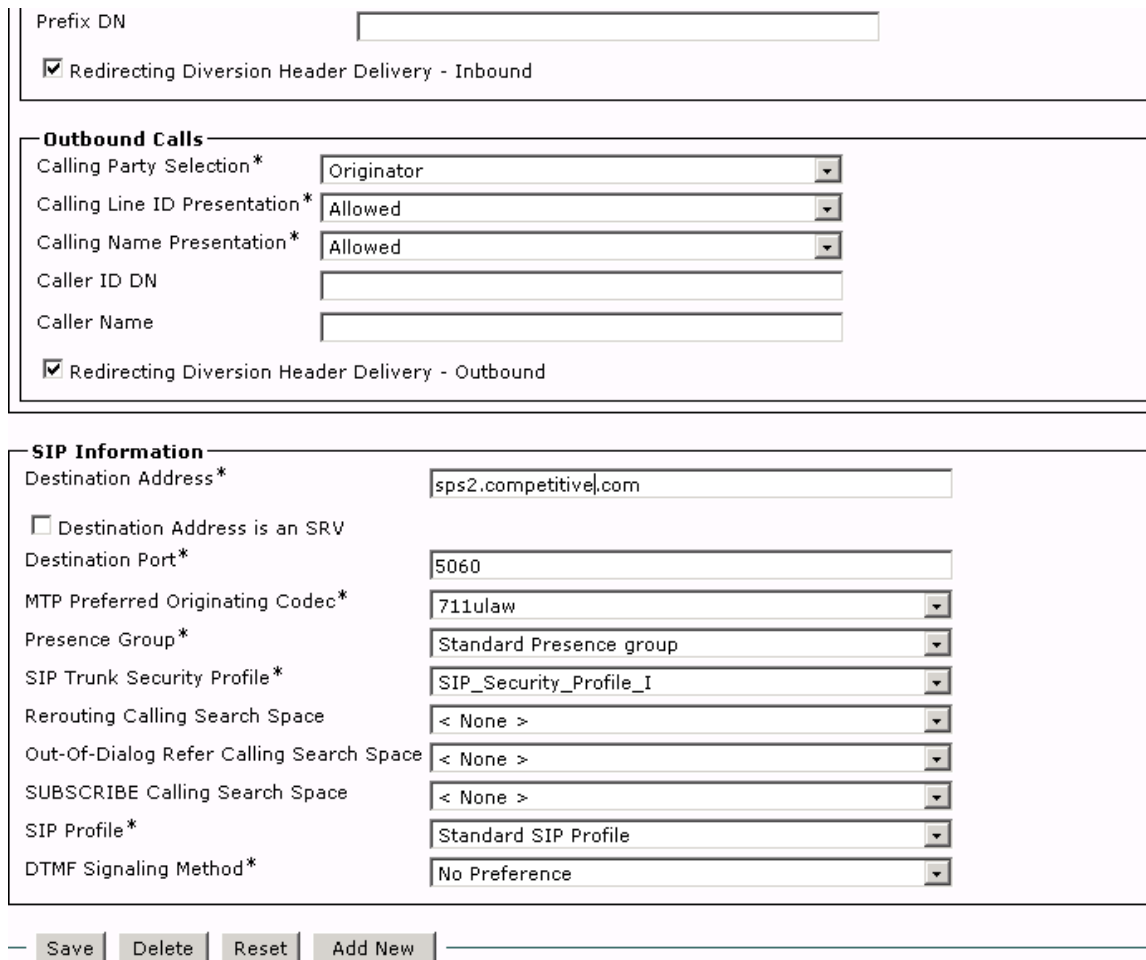
<b>Device Information</b>	
Product:	SIP Trunk
Device Protocol:	SIP
Device Name *	<input type="text" value="SIP_to_CS1K2"/>
Description	<input type="text" value="SIP_to_CS1K2"/>
Device Pool *	<input type="text" value="G711_Pool"/>
Common Device Configuration	<input type="text" value="MigratedCommonDeviceConfig3"/>
Call Classification *	<input type="text" value="Use System Default"/>
Media Resource Group List	<input type="text" value="sw_mrg1"/>
Location *	<input type="text" value="Hub_None"/>
AAR Group	<input type="text" value=" &lt; None &gt;"/>
Packet Capture Mode *	<input type="text" value="None"/>
Packet Capture Duration	<input type="text" value="0"/>
<input checked="" type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Unattended Port	

<b>Multilevel Precedence and Preemption (MLPP) Information</b>	
MLPP Domain	<input type="text" value=" &lt; None &gt;"/>

<b>Call Routing Information</b>	
<b>Inbound Calls</b>	
Significant Digits *	<input type="text" value="All"/>
Connected Line ID Presentation *	<input type="text" value="Allowed"/>
Connected Name Presentation *	<input type="text" value="Allowed"/>
Calling Search Space	<input type="text" value=" &lt; None &gt;"/>
AAR Calling Search Space	<input type="text" value=" &lt; None &gt;"/>
Prefix DN	<input type="text"/>

Note: This is a proprietary screenshot from Cisco Unified Communications Manager 6.0

SIP Trunk Configuration continued.



Prefix DN

☒ Redirecting Diversion Header Delivery - Inbound

---

**Outbound Calls**

Calling Party Selection\*

Calling Line ID Presentation\*

Calling Name Presentation\*

Caller ID DN

Caller Name

☒ Redirecting Diversion Header Delivery - Outbound

---

**SIP Information**

Destination Address\*

☐ Destination Address is an SRV

Destination Port\*

MTP Preferred Originating Codec\*

Presence Group\*

SIP Trunk Security Profile\*

Rerouting Calling Search Space

Out-Of-Dialog Refer Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile\*

DTMF Signaling Method\*

—     —

Note: This is a proprietary screenshot from Cisco Unified Communications Manager 6.0

Put **competitive.com** in the Destination Address field if you are using **com** as the network domain.

## **Call Routing Configuration on Cisco Unified Communications Manager**


A route pattern comprises a string of digits (an address) and a set of associated digit manipulations that route calls to a route list or a gateway. Route patterns provide flexibility in network design. Please enter the correct digits for the destination and associate with the Gateway/Route from the drop down menu.

Dialing Plan (Cisco Route Patterns under Call Routing drop down menu)

9.4XXX → SIP → CS1K

9.5XXXX → SIP → CallPilot

From Cisco Unified Communications Manager Administration Window drop down menu.  
Select:- Call Routing / Route Hunt / Route pattern / Add New.

**Status**
  
 Status: Ready

---

**Pattern Definition**
  
 Route Pattern\* 
  
 Route Partition 
  
 Description 
  
 Numbering Plan 
  
 Route Filter 
  
 MLPP Precedence\* 
  
 Gateway/Route List\*  [\(Edit\)](#)
  
 Route Option
 

☒ Route this pattern
 

☐ Block this pattern

  
 Call Classification\* 
  
☐ Allow Device Override
 ☒ Provide Outside Dial Tone
 ☐ Allow Overlap Sending
 ☒ Urgent Priority
   
☐ Require Forced Authorization Code
   
 Authorization Level\* 
  
☐ Require Client Matter Code

---

**Calling Party Transformations**
  
☐ Use Calling Party's External Phone Number Mask
   
 Calling Party Transform Mask 
  
 Prefix Digits (Outgoing Calls) 
  
 Calling Line ID Presentation\* 
  
 Calling Name Presentation\*

Note: This is a proprietary screenshot from Cisco Unified Communications Manager 6.0

**Calling Party Transformations**☐ Use Calling Party's External Phone Number MaskCalling Party Transform Mask Prefix Digits (Outgoing Calls) Calling Line ID Presentation\* Calling Name Presentation\* **Connected Party Transformations**Connected Line ID Presentation\* Connected Name Presentation\* **Called Party Transformations**Discard Digits Called Party Transform Mask Prefix Digits (Outgoing Calls) **ISDN Network-Specific Facilities Information Element**Network Service Protocol Carrier Identification Code 

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value="&lt; Not Exist &gt;"/>	<input type="text"/>

Save

Delete

Copy

Add New

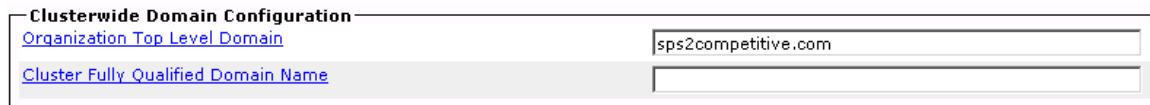
Note: This is a proprietary screenshot from Cisco Unified Communications Manager 6.0

### **Cluster Wide Domain Name Setup on Cisco Unified Communications Manager**

From Cisco Unified Communications Manager Administration Window drop down menu. Select:- System / Enterprise Parameters.

Enter Domain name.

Note: For SIP Trunks the Domain name must match the top level Domain in Nortel SPS / NRS. The name must be in “doted” notation, e.g. nortel.com, university.org. Stand alone name e.g. nortel or university will be rejected by Cisco Unified Communications Manager.



Clusterwide Domain Configuration	
Organization Top Level Domain	sps2competitive.com
Cluster Fully Qualified Domain Name	

Note: This is a proprietary screenshot from Cisco Unified Communications Manager 6.0

Put **competitive.com** in the Organization Top Level Domain field if you are using **com** as the network domain.

## Nortel Signaling Server Configuration Sequence.

### **Zone Configuration on Nortel Communication server 1000**

Zones are used to specify the bandwidth that is used for audio and video calls within a region and between existing regions.

IP Telephony zones corresponds to commands in LD 117, Ethernet Management. Use the default and do not change any settings.

From Nortel Element Manager main window  
Select:- System / IP Network / Zones

#### **Zones**

##### **Maintenance**

- Maintenance Commands for Zones (LD 117)

##### **Configuration**

- Configuration Spreadsheet

Please Choose the **Zone 1** to Add

##### **- Zone 0**

- Zone Basic Property and Bandwidth Management
- Adaptive Network Bandwidth Management and CAC
- Alternate Routing for Calls between IP Stations
- Branch Office Dialing Plan and Access Codes
- Branch Office Time Difference and Daylight Saving Time Property

Select:- System / IP Network / Zones / Zone 0 / Zone Basic Property and Bandwidth Management

Managing: **10.10.132.134**

System » IP Network » Zones » Zone 0 » Zone Basic Property and Bandwidth Management

#### **Zone Basic Property and Bandwidth Management**

Input Description	Input Value
Zone Number (ZONE):	<input type="text" value="0"/>
Intrazone Bandwidth (INTRA_BW):	<input type="text" value="1000000"/>
Intrazone Strategy (INTRA_STGY):	<input type="text" value="Best Quality (BQ)"/>
Interzone Bandwidth (INTER_BW):	<input type="text" value="1000000"/>
Interzone Strategy (INTER_STGY):	<input type="text" value="Best Quality (BQ)"/>
Resource Type (RES_TYPE):	<input type="text" value="Shared (SHARED)"/>
Zone Intent (ZBRN):	<input type="text" value="MO (MO)"/>
Description (ZDES):	<input type="text"/>

## **Node IP Configuration on Nortel Communication server 1000**

View basic information on existing Nodes here. You can add a node, or select a node to edit or delete a Node.

From Nortel Element Manager main window

Select:- System / IP Network / Nodes: Servers, Media Cards / Node Configuration.

Managing **10.10.132.134**  
System > IP Network > Node Configuration

### **Node Configuration**

New Node

- Node: 434	Node IP: <b>10.10.132.176</b>	<input type="button" value="Edit..."/>	<input type="button" value="Transfer / Status"/>	<input type="button" value="Delete"/>
Telephony LAN (TLAN) IP address		<input type="text" value="TN"/>		
<b>Signaling Server</b>				
10.10.132.175				

## IP Telephony Node Configuration Continued.

Managing: **10.10.132.134**

System » IP Network » [Node Configuration](#) » IP Telephony: Node ID 434 » Edit

### Edit

**- IP Telephony Node**

**Node ID 434**

**Telephony LAN (TLAN) Node IP address**  \*

**Embedded LAN (ELAN) gateway IP address**

**Embedded LAN (ELAN) subnet mask**

**Voice LAN (TLAN) subnet mask**

- + VGW and IP phone codec profile
- + QoS
- + LAN configuration
- + SNTP
- + Virtual Trunk Network Health Monitor configuration
- + H323 GW Settings
- + Firmware
- + SIP GW Settings
- + SIP URI Map
- + SIP CD Services
- + SIP CTI Services
- + Cards
- + Signaling Servers

\* *Mandatory fields of current configuration*





### **VGW and IP Phone Codec Configuration on Nortel Communication server 1000**

View basic information on existing codecs here. G711 is the default codec with 20ms payload, supported payloads are 30, 20, 10 ms. You can select a codec to edit its properties. Cisco Unified Communications Manager with MTP enabled requires G711 codec.

From Nortel Element Manager main window

Select:- System / IP Network / Nodes: Servers, Media Cards / VGW and IP Phone Codec Profile.

Managing **10.10.132.134**System » IP Network » Node Configuration » IP Telephony: Node ID 434 » Edit**Edit**

+ IP Telephony Node

- VGW and IP phone codec profile

Enable Echo canceller ☒

Echo canceller tail delay  ( milliseconds )

Voice activity detection threshold  ( -20 - +10 DBM )

Idle noise level  ( -327 - +327 DBM )

DTMF Tone detection ☒

Enable V.21 FAX tone detection ☐

FAX maximum rate  ( bps )

FAX playout nominal delay  ( 0 - 300 milliseconds )

FAX no activity timeout  ( 10 - 32000 milliseconds )

FAX packet size

- Codec G711

Select ☒

Codec Name G711

Voice payload size  ( ms/frame )

Voice playout (jitter buffer) nominal delay

Modifications may cause changes to maximal delay settings

Voice playout (jitter buffer) maximum delay

- Codec G729A

Select ☒

Codec Name G729A

Voice payload size  ( ms/frame )

Voice playout (jitter buffer) nominal delay

Modifications may cause changes to maximal delay settings

Voice playout (jitter buffer) maximum delay

VAD ☐

+ Codec G723.1

Select ☐

+ Codec T38 FAX

Select ☒

**QoS Configuration on Nortel Communication server 1000**

View QoS information here, Changes made to QoS will not effect Cisco clients.

From Nortel Element Manager main window

Select:- System / IP Network / Nodes: Servers, Media Cards / QoS.

Managing: **10.10.132.134**System » IP Network » [Node Configuration](#) » IP Telephony: Node ID 434 » Edit**Edit**

<input type="button" value="Save and Transfer"/>		<input type="button" value="Cancel"/>
<b>+ IP Telephony Node</b>		
<b>+ VGW and IP phone codec profile</b>		
<b>- QoS</b>		
Diffserv Codepoint(DSCP) Control packets	<input type="text" value="40"/>	( 0 - 63 )
Diffserv Codepoint(DSCP) Voice packets	<input type="text" value="46"/>	( 0 - 63 )
Enable 802.1Q support	<input type="checkbox"/>	
802.1Q Bits value (802.1p)	<input type="text" value="6"/>	( 0 - 7 )

**LAN Configuration on Nortel Communication server 1000**

View or Change LAN information. Enter Call Server IP address and leave all settings to default.

From Nortel Element Manager main window

Select:- System / IP Network / Nodes: Servers, Media Cards / LAN

Managing: **10.10.132.134**System » IP Network » [Node Configuration](#) » IP Telephony: Node ID 434 » Edit**Edit**

Save and Transfer		Cancel
+ IP Telephony Node		
+ VGW and IP phone codec profile		
+ QoS		
- LAN configuration		
Embedded LAN (ELAN) configuration		
Call server IP address	10.10.132.134	
Unistern Signaling port	15000	
Broadcast port	15001	( 1024 - 65535 )
Telephony LAN (TLAN) configuration		
Unistern Signaling port	5000	
RTP/RTCP Starting port	5200	( 1024 - 65535 )
Embedded LAN (ELAN) Routes	Add	
Host Table	Add	
DNS Servers		
Primary DNS Server IP address	0.0.0.0	
Alternate DNS Server 1 IP address	0.0.0.0	
Alternate DNS Server2 IP address	0.0.0.0	

**SIP GW Configuration on Nortel Communication server 1000**

View or Change SIP GW settings here. Since this is a co-resident the primary proxy redirect server has the same IP address as Signaling Server.

Note: Cisco Unified Communications Manager will use the Node IP address to communicate via SIP trunk.

From Nortel Element Manager main window  
 Select:- System / IP Network / Nodes: Servers, Media Cards / SIP GW Settings

Managing: **10.10.132.134**

System > IP Network > Node Configuration > IP Telephony: Node ID 434 > Edit

## Edit

<input type="button" value="Save and Transfer"/> <input type="button" value="Cancel"/>	
<div> <div>+ IP Telephony Node</div> <div>+ VGW and IP phone codec profile</div> <div>+ QoS</div> <div>+ LAN configuration</div> <div>+ SNTP</div> <div>+ Virtual Trunk Network Health Monitor configuration</div> <div>+ H323 GW Settings</div> <div>+ Firmware</div> <div>- SIP GW Settings</div> <div>TLS Security</div> </div>	
Security Policy	Security Disabled
TLS Security Port	5061 ( 1 - 65535 )
Client Authentication	<input type="checkbox"/>
Re-negotiation	<input type="checkbox"/>
X.509 Certificate Authentication	<input type="checkbox"/>
Primary Proxy or Re-direct Server	
Primary Proxy or Redirect (TLAN) IP address	10.10.132.164
Port	5060
Supports Registration	<input checked="" type="checkbox"/>
Primary CDS Proxy or Re-direct server flag	<input type="checkbox"/>
Transport Protocol	TCP
Secondary Proxy or Re-direct Server	
Secondary Proxy or Redirect (TLAN) IP address	0.0.0.0
Port	5060
Supports Registration	<input type="checkbox"/>
Secondary CDS Proxy or Re-direct server flag	<input type="checkbox"/>
Transport Protocol	TCP
CLID Parameters	
Country Code (CCC)	0
Area Code (AreaCode)	5061
# Digits to Strip	0
Prefix to Insert	0
Subscriber Number (SN)	0
National Number (NN)	0
International number	
<i>Note: The NPA in North America</i> Format of CLID +<CCC><AreaCode><SN> +<CCC><NN> +<International number>	

## Signalling Server Configuration on Nortel Communication server 1000

Configure Signaling Server settings on this page.

Note: SIP Domain Name (encircled below) must match the Service Domain in Nortel SPS and Organization Top Level Domain in Cisco Unified Communications Manager Enterprise Parameters.

From Nortel Element Manager main window

Select:- System / IP Network / Nodes: Servers, Media Cards / SIP GW Settings

Managing: **10.10.132.134**

System » IP Network » Node Configuration » IP Telephony: Node ID 434 » Edit

## Edit

Save and Transfer

Cancel

+ IP Telephony Node

+ VGW and IP phone codec profile

+ QoS

+ LAN configuration

+ SNTP

+ Virtual Trunk Network Health Monitor configuration

+ H323 GW Settings

+ Firmware

+ SIP GW Settings

+ SIP URI Map

+ SIP CD Services

+ SIP CTI Services

+ Cards

- Signaling Servers

- Signaling Server **10.10.132.141** Properties

Add

Add

Remove

Role Leader

Type CPPM

Embedded LAN (ELAN) IP address **10.10.132.141**

Embedded LAN (ELAN) MAC address **00:02:b3:f1:b4:c4**

Telephony LAN (TLAN) IP address **10.10.132.175**

Telephony LAN (TLAN) gateway IP address **10.10.132.161**

Hostname **CS2**

H323 ID **CS2**

Enable Line TPS ☒

Enable IP Peer Gateway (Virtual Trunk TPS) **H.323 and SIP**

If Telephony LAN (TLAN) IP address and Telephony LAN (TLAN) gateway IP address are enabled, then the TLAN must be enabled when Line TPS or IP Peer Gateway is enabled, then the TLAN must be enabled.

Enable SIP Proxy / Redirect Server ☒

Local SIP TCP/UDP Port to Listen to **5060**

SIP Domain name **sps2.competitive.com**

SIP Gateway Endpoint Name **CS2**

SIP Gateway Authentication Password **\*\*\*\*\***

Enable Gatekeeper ☒

Network Routing Service Role **Primary**

## (MGC) Media Gateway Controller Configuration on Nortel Communication server 1000

Configure MGC if you need conferencing or non-IP clients to interop with CCM. Enter correct IP address for MGC and daughter board and use default settings for the rest.

From Nortel Element Manager main window  
Select:- System / IP Network / Media Gateways

Managing: **10.10.132.134**

System » IP Network » Media Gateways » IPMG 0 0 Media Gateway Controller (MGC) Configuration

## IPMG 0 0 Media Gateway Controller (MGC) Configuration

<b>- Media Gateway Controller</b>	
Hostname	MGC *
Management LAN (ELAN) IP address	10.10.132.142 *
Management LAN (ELAN) gateway IP address	10.10.132.129 *
Management LAN (ELAN) subnet mask	255.255.255.224
Voice LAN (TLAN) IP address	10.10.132.180 *
Voice LAN (TLAN) gateway IP address	10.10.132.161 *
Voice LAN (TLAN) subnet mask	255.255.255.224
<b>- DSP Daughterboard 1</b>	
Type of the DSP Daughterboard	NODB
Voice LAN (TLAN) IP address	10.10.132.178 *
Voice LAN (TLAN) gateway IP address	10.10.132.161 *
Voice LAN (TLAN) subnet mask	255.255.255.224
Hostname	DB1 *
<b>- DSP Daughterboard 2</b>	
Type of the DSP Daughterboard	DB32
Voice LAN (TLAN) IP address	10.10.132.179 *
Voice LAN (TLAN) gateway IP address	10.10.132.161 *
Voice LAN (TLAN) subnet mask	255.255.255.224
Hostname	DB2 *
<b>+ VGW and IP phone codec profile</b>	
<b>+ QoS</b>	
<b>+ LAN configuration</b>	
Submit	Cancel
VGW Channels	

\* Mandatory fields of current configuration

## Nortel (SPS) SIP Proxy Server Configuration Sequence.

## **(SPS) SIP proxy Server System Wide Configuration on Communication server 1000**

The (SPS) SIP Proxy Server uses a basic SIP structure for its configuration, which is applicable for SIP, H.323, and Network Connection Server (NCS) call completion. This structure is the basis of the single network dialing/numbering plan.

SPS/NRS System Wide Settings, Please use default values.

Managing: 10.10.132.164

System » [System Wide Settings](#)

### **System Wide Settings**

SIP registration time to live timer:	<input type="text" value="300"/>	(30-3600 Seconds)
H.323 gatekeeper registration time to live timer:	<input type="text" value="300"/>	(30-3600 Seconds)
H.323 alias name:	<input type="text"/>	*
Auto backup time:	<input type="text" value="23:49"/>	(HH:MM)
Auto backup to secure FTP site enabled:	<input type="checkbox"/>	
Auto backup to secure FTP site's IP address:	<input type="text"/>	
Auto backup secure FTP site's path:	<input type="text"/>	
Auto backup secure FTP user name:	<input type="text"/>	
Auto backup secure FTP password:	<input type="text"/>	

\* Required value.



## **(SPS) SIP proxy Server Configuration on Communication server 1000**

Managing: 10.10.132.164

System » [NRS Server](#)

### NRS Server

#### Service Status

<input type="checkbox"/>	Service Name	Service Status
1 <input type="checkbox"/>	SIP Proxy Server (SPS)	In service
2 <input type="checkbox"/>	Gatekeeper (GK)	In service
3 <input type="checkbox"/>	Network Connection Server (NCS)	In service

#### Server Configuration

##### NRS Setting

**Host name** sps2  
**Primary TLAN IP address** 10.10.132.164  
**Secondary TLAN IP address** 0.0.0.0  
**Secondary server host name** SecondaryHostName  
**Control priority** 40  
**Server mate communication port** 5005  
**Realm name** realmName  
**Server role** Primary

##### H.323 Gatekeeper Settings

**Location request (LRQ) response timeout** 3

##### SIP Server Settings

**Mode** proxy  
**Public name for non-trusted networks** unknown  
**Public number for non-trusted networks** 000-000  
**UDP Transport enabled** ☒  
**Primary server UDP IP** 10.10.132.164  
**Primary server UDP port** 5060  
**Secondary server UDP IP** 0.0.0.0  
**Secondary server UDP port** 5060  
**UDP maximum transmission unit (MTU)** 1500  
**TCP Transport enabled** ☒  
**Primary server TCP IP** 10.10.132.164

SPS/NRS Configuration continued.

SPS/NRS System Wide Settings, use defaults, no change required.

Primary server TCP port 5060  
Secondary server TCP IP 0.0.0.0  
Secondary server TCP port 5060  
TCP maximum transmission unit (MTU) 1500  
TLS Transport enabled ☒  
Primary server TLS IP 10.10.132.164  
Primary server TLS port 5061  
Secondary server TLS IP 0.0.0.0  
Secondary server TLS port 5061  
TLS maximum transmission unit (MTU) 1500  
Transport Layer Security (TLS) Settings  
Maximum session cache 2048000

Session cache timeout 600  
Renegotiation in byte 2048000  
X509 Certificate authentication ☒  
Client authentication ☐  
Network Connection Server (NCS) Settings  
Primary NCS port 16500  
Secondary NCS port 16500  
Primary NCS timeout 10

SPS / SIP Configuration continued.

### **Service Domain Configuration on Nortel Communication server 1000**

Note: Service Domain Name (encircled below) must match the SIP Domain in Nortel SS and Organization Top Level Domain in Cisco Unified Communications Manager Enterprise Parameters.

**Domains**  
Domains establish the basic structure of your converged network, defined by Service domains, L1 (UDP) and L0 (CDP) domains.

Service Domains (1)		L1 Domains (UDP) (1)	L0 Domains (CDP) (1)			Refresh
<input type="checkbox"/> ID ^	Description	# of L1 Domains	# of L0 Domains	# of Gateway Endpoints		
1 <u>sps2.competitive.com</u>	Cisco	1	1	2		

1 - 1 of 1 Service Domain(s) First | Previous | Next | Last

### **L1 (UDP) Domain Configuration on Nortel Communication server 1000**

Note: Create UDP Domain under Service Domain Name that match the Top Level Domain in Cisco Unified Communications Manager Enterprise Parameters.

Domains

Domains establish the basic structure of your converged network, defined by Service domains, L1 (UDP) and L0 (CDP) domains.

Service Domains (1)

L1 Domains (UDP) (1)

L0 Domains (CDP) (1)

Filter by Domain: All service domains

ID

▲

Description

# of L0 Domains

# of Gateway Endpoints

# of Routing Entries

Context

1

udp

1

2

2

sps2.competitive.com

1 - 1 of 1 L1 Domain(s)

First

Previous

Next

Last

## **L0 (CDP) Domain Configuration on Nortel Communication server 1000**

Note: Create UDP/CDP Domain under Service Domain Name that matches the Top Level Domain in Cisco Unified Communications Manager Enterprise Parameters.

**Domains**  
Domains establish the basic structure of your converged network, defined by Service domains, L1 (UDP) and L0 (CDP) domains.

**Service Domains (1)**    **L1 Domains (UDP) (1)**    **L0 Domains (CDP) (1)**

Filter by Domain:  /

[Refresh](#)

<input type="checkbox"/> ID ▲	Description	# of Gateway Endpoints	# of Routing Entries	Context
1 <input type="checkbox"/> cdp		2	2	sps2.competitive.com / udp

1 - 1 of 1 L0 Domain(s)

[First](#) | [Previous](#) | [Next](#) | [Last](#)

## **SIP Gateway Endpoint Configuration on Nortel Communication server 1000**

Enter Endpoint name, description, Enter the IP address of Cisco Callmanger. Select Static SIP endpoint, use TCP for transport and default SIP port of 5060.

Managing: ☒ Active database 10.10.132.164  
☐ Standby database [Numbering Plan > Endpoints > Gateway Endpoint](#)

Edit Gateway Endpoint ( sps2.competitive.com / udp / cdp )

End point name:	<input type="text" value="Cisco_50_SIP_H323"/> *
Description:	<input type="text"/>
Trust Node:	<input checked="" type="checkbox"/>
Tandem gateway endpoint name:	<input type="text" value="Not configured"/>
Endpoint authentication enabled:	<input type="text" value="Authentication off"/>
Authentication password:	<input type="text"/>
E.164 country code:	<input type="text"/>
E.164 area code:	<input type="text"/>
E.164 international dialing access code:	<input type="text"/>
E.164 international dialing code length:	<input type="text"/>
E.164 national dialing access code:	<input type="text"/>
E.164 national dialing code length:	<input type="text"/>
E.164 local (subscriber) dialing access code:	<input type="text"/>
E.164 local (subscriber) dialing code length:	<input type="text"/>
Private L1 domain (UDP location) dialing access code:	<input type="text"/>
Private L1 domain (UDP location) dialing code length:	<input type="text"/>
Private Special number 1:	<input type="text"/>
Private Special number 1 dialing code length:	<input type="text"/>
Private Special number 2:	<input type="text"/>
Private Special number 2 dialing code length:	<input type="text"/>
Static endpoint address type:	<input type="text" value="IP version 4"/>
Static endpoint address:	<input type="text" value="10.10.133.100"/>
H.323 support:	<input type="text" value="Not RAS H.323 endpoint"/>
SIP support:	<input type="text" value="Static SIP endpoint"/>
SIP TCP transport enabled:	<input checked="" type="checkbox"/>
SIP TCP port:	<input type="text" value="5060"/>
SIP UDP transport enabled:	<input checked="" type="checkbox"/>
SIP UDP port:	<input type="text" value="5060"/>
SIP TLS transport enabled:	<input type="checkbox"/>
SIP TLS port:	<input type="text" value="5061"/>
Persistent TCP support enabled:	<input checked="" type="checkbox"/>
End to end security support:	<input type="checkbox"/>
Network Connection Server enabled:	<input type="checkbox"/>
Redundancy enabled:	<input type="text" value="Not Configured"/>
Main endpoint name:	<input type="text" value="Not configured"/>
Redundant endpoint name:	<input type="text" value="Not configured"/>

## **SIP Routing Entries Configuration on Nortel Communication server 1000**

36XXX is a SIP route to CCM

Managing: ☒ Active database 10.10.132.164  
☐ Standby database [Numbering Plan» Routes](#)

### Search for Routing Entries

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

DN Prefix:  DN Type:   
 Limit results to Domain:  /  /   
 Endpoint Name:

Results per page:

Routing Entries (4)		Default Routes (0)			
Routing test...					Refresh
<input type="checkbox"/> DN Prefix ▲	DN Type	Route Cost	SIP URI Phone Context	Context	
1 <input type="checkbox"/> 36	Private level 0 regional (CDP steering code)	1	cdp.udp	sps2.competitive.com / udp / cdp / Cisco_50_SIP_H323	
2 <input type="checkbox"/> 40	Private level 0 regional (CDP steering code)	1	cdp.udp	sps2.competitive.com / udp / cdp / CS2	
3 <input type="checkbox"/> 47	Private level 0 regional (CDP steering code)	1	cdp.udp	sps2.competitive.com / udp / cdp / Cisco_50_SIP_H323	
4 <input type="checkbox"/> 70	Private level 0 regional (CDP steering code)	1	cdp.udp	sps2.competitive.com / udp / cdp / Clab_Avaya_H323	

1 - 4 of 4 Routing Entry(ies)

First| Previous| Next| Last

## Nortel Call Server Configuration Sequence.

### IP D-channel Configuration on Nortel Communication server 1000

Use EM to configure D-channel

CS 1000 ELEMENT MANAGER	
- Basic Configuration	
Input Description	Input Value
Action Device And Number (ADAN) (TYPE)	DCH
D channel Card Type (CTYP)	DCIP
Designator (DES)	dch
Recovery to Primary (RCVP)	<input type="checkbox"/>
PRI loop number for Backup D-channel (BCHL)	
User (USR)	Integrated Services Signaling Link Dedicated (ISLD)
Interface type for D-channel (IFC)	Meridian Meridian1 (SL1)
Country (CNTY)	ETS 300=102 basic protocol (ETSI)
D-Channel PRI loop number (DCHL)	
Primary Rate Interface (PRI)	<input type="text"/> more PRI
Secondary PRI2 loops (PRI2)	
Meridian 1 node type (SIDE)	Slave to the controller (USR)
Release ID of the switch at the far end (RLS)	25
Central Office switch type (CO_TYPE)	100% compatible with Bellcore standard (STD)
Integrated Services Signaling Link Maximum (ISLM)	4000 Range: 1 - 4000
Signaling Server Resource Capacity (SSRC)	1800 Range: 0 - 4000
- Basic options (BSCOPT)	
Primary D-channel for a backup DCH (PDCH)	Range: 0 - 254
- PINX customer number (PINX_CUST)	
- Progress signal (PROG)	
- Calling Line Identification (CLID)	
- Output request Buffers (OTBF)	32
- D-channel transmission Rate (DRAT)	56 kb/s when LCMT is AMI (56K)
- Channel Negotiation option (CNEG)	No alternative acceptable, exclusive. (1)
- Remote Capabilities (RCAP)	Edit
- - Change protocol timer value (TIMR)	
- How long Meridian 1 to wait for the response message when the OSIG outgoing call is in the U3 state (T310)	120
- Variable timer for received disconnect message on incoming calls (INC_T306)	2 Range: 0 - 240
- Variable timer for received disconnect message (OUT_T306)	30 Range: 0 - 240
- B channel Service messaging. (BSRV)	<input type="checkbox"/>

## D-channel Configuration continued.

## - Advanced options (ADVOPT)

- Layer 3 call control message count per 5 second time interval (ISDN\_MCNT)  Range: 60 - 350
- Number of Status Enquiry Messages sent within 128 ms (SEMT)
- Map channel number to timeslots on a PRI2 loop (QCHID) ☒

## - H323 Overlap Signaling Settings (H323)

- Overlap Receiving (OVLK) ☐
- Overlap Sending (OVLS) ☐
- Overlap Timer (OVLTI)
- Multilocation Business Group Allowed (MBGA) ☐
- Network Attendant Service Allowed (NASA) ☒

## - Link Access Protocol for D-channel (LAPD)

- Interface guard Timer or DCHL only (T23)
- Retransmission Timer (T200)
- Maximum Time allowed without frames being exchanged (T203)
- Maximum Number of retransmissions (N200)
- Maximum Number of octets in information element (N201)  Range: 4 - 260
- Maximum number of outstanding unacknowledged frames (K)
- Maximum number of status inquiries when remote is busy (N2X4)

## - Feature Packages

## - Digital Private Network Signaling System 1

Package: 123

Input Description	Input Value
Digital Private Network Signaling (DPNS)	<input type="checkbox"/>

## - Virtual Network Services

Package: 183

Input Description	Input Value
Virtual Network Services Network Signalling option (VNSIG)	<input type="checkbox"/>
Virtual Network Services Maximum (VNSM)	<input type="text" value="1"/> Range: 1 - 300
Virtual Network Services Customer number (VNSC)	<input type="text" value="0"/>
Virtual Network Services Private Network Identifier (VNSPI)	<input type="text" value="0"/> Range: 0 - 32700
Virtual Network Services Network Call Party Name Display (VCNA)	<input type="checkbox"/>
Virtual Network Services Network Call Redirection (VCRD)	<input type="checkbox"/>



You can also use CLI to configure D-Channel: Use LD 17 to configure D-channel signalling between Call Server and Signalling Server.

```
ADAN    DCH 10
CTYP DCIP
DES vtrk
USR ISLD
ISLM 4000
SSRC 1800
OTBF 32
NASA NO
IFC SL1
CNEG 1
RLS ID 25
RCAP ND3
MBGA NO
H323
OVLN NO
OVLS NO
```

## Super-Loop Configuration on Nortel Communication server 1000

Use EM to configure super-loops

### CS 1000 ELEMENT MANAGER

Managing: 10.10.132.134

System » Core Equipment » Superloops » Superloops 0 Property Configuration

### Superloops 0 Property Configuration

Input Description	Input Value
Superloop (SUPL)	<input type="text" value="000"/>
Superloop Type (SUPT)	<input type="text" value="IPMG"/>

#### Shelf 0

Input Description	Input Value
Zone number (ZONE0)	<input type="text" value="000"/> Range: 0 - 255
IP based Media Gateway 0 (IPMG0)	<input type="text" value="0"/>
ELAN IP address (IPR0)	<input type="text" value="10.10.132.142"/>
IPMG Type (TYP0)	<input type="text" value="MGC"/>
ELAN Passthrough Port (CE0)	<input type="text" value="CE"/>
Faceplate ELAN Port (E10)	<input type="text" value="1E"/>
Backplane ELAN Connection (E0)	<input type="text" value="E"/>
TLAN Passthrough Port (CT0)	<input type="text" value="CT"/>
Faceplate TLAN Port (T20)	<input type="text" value="2T"/>
Backplane TLAN Connection (T0)	<input type="text" value="T"/>

#### Shelf 1

Input Description	Input Value
Zone Number (ZONE1)	<input type="text"/> Range: 0 - 255
IP based Media Gateway 1 (IPMG1)	<input type="text" value="1"/>
ELAN IP Address (IPR1)	<input type="text"/>
IPMG Type (TYP1)	<input type="text" value="MGC"/>
ELAN Passthrough Port (CE1)	<input type="text" value="CE"/>
Faceplate ELAN Port (E11)	<input type="text" value="1E"/>
Backplane ELAN Connection (E1)	<input type="text" value="1ELAN"/>
TLAN Passthrough Port (CT1)	<input type="text" value="CT"/>
Faceplate TLAN Port (T21)	<input type="text" value="2T"/>
Backplane TLAN Connection (T1)	<input type="text" value="2TLAN"/>

Super-Loop Configuration continued.

### Superloops 96 Property Configuration

Input Description	Input Value
Superloop (SUPL)	096
Superloop Type (SUPT)	Virtual

### Superloops 100 Property Configuration

Input Description	Input Value
Superloop (SUPL)	100
Superloop Type (SUPT)	Virtual

You can also use CLI to configure Super-Loops: Use LD 97 to configure Super-Loops, Super-Loops are required for Virtual Trunks.

SUPL SUPT SLOT XPEC0 XPEC1 IPMG ZONE0/1 IPR0/1

```

000 IPMG ---- - - - - - 001 000 10.10.132.142 (SUPL for MGC)
096 ---- VIRTUAL - - - - - --- --- (Virtual SUPL)
100 ---- VIRTUAL - - - - - --- --- (Virtual SUPL)

```

## SIP Routes and Trunks Configuration on Nortel Communication server 1000

Use EM to configure Routes and Trunks

### CS 1000 ELEMENT MANAGER

Managing: 10.10.132.134

Routes and Trunks » [Routes and Trunks](#) » Customer 0, Route 11 Property Configuration

#### Customer 0, Route 11 Property Configuration

##### - Basic Configuration

Input Description	Input Value
Route Data Block (RDB) (TYPE)	RDB
Customer number (CUST)	00
Route Number (ROUT)	11
Designator field for trunk (DES)	SIP
Trunk Type (TKTP)	TIE
Incoming and Outgoing trunk (ICOG)	Incoming and Outgoing (IAO)
Access Code for the trunk route (ACOD)	8011
Trunk type M911P (M911P)	<input type="checkbox"/>
The route is for a virtual trunk route (VTRK)	<input checked="" type="checkbox"/>
- Zone for codec selection and bandwidth management (ZONE)	003 <span style="color: green;">Range: 0 - 255</span>
- Node ID of signaling server of this route (NODE)	434 <span style="color: green;">Range: 0 - 9999</span>
- Protocol ID for the route (PCID)	SIP (SIP)
- Print Correlation ID in CDR for the route (CRID)	<input type="checkbox"/>
Integrated Services Digital Network option (ISDN)	<input checked="" type="checkbox"/>
- Mode of operation (MODE)	Route uses ISDN Signaling Link (ISLD)
- D channel number (DCH)	10 <span style="color: green;">Range: 0 - 254</span>
- Interface type for route (IFC)	Meridian M1 (SL1)
- Private Network Identifier (PNI)	00001 <span style="color: green;">Range: 0 - 32700</span>
- Network Calling Name Allowed (NCNA)	<input checked="" type="checkbox"/>
- Network Call Redirection (NCRD)	<input checked="" type="checkbox"/>
- Trunk Route Optimization (TRO)	<input type="checkbox"/>
- Recognition of DTI2 ABCD FALT signal for ISL (FALT)	<input type="checkbox"/>
- Channel Type (CHTY)	B-channel (BCH)
- Call Type for outgoing direct dialed TIE route (CTYP)	Unknown Call type (UKWN)
- Insert ESN Access Code (INAC)	<input checked="" type="checkbox"/>
- Integrated Service Access Route (ISAR)	<input type="checkbox"/>
- Display of Access Prefix on CLID (DAPC)	<input type="checkbox"/>
- Mobile Extension Route (MBXR)	<input type="checkbox"/>

## Routes and Trunks Configuration continued.

## - Basic Route Options

Input Description	Input Value
Billing Number Required (BILN)	<input type="checkbox"/>
Call Detail Recording (CDR)	<input type="checkbox"/>
Controls or timers (CNTL)	<input type="checkbox"/>
Conventional (Tie trunk only) (CNVT)	<input type="checkbox"/>
Incoming DID Digit Conversion on this route (IDC)	<input type="checkbox"/>
Multifrequency Compelled or MFC Signaling (MFC)	No MFC (NO) ▾
Process Notification Networked Calls (PNNC)	<input type="checkbox"/>

## - Network Options

Input Description	Input Value
Electronic Switched Network pad control (ESN)	<input type="checkbox"/>
Signaling arrangement (SIGO)	DTI data calls plus all other types (ESN5) ▾
Route Class (RCLS)	Route Class marked as external (EXT) ▾
Off-Hook Queuing (OHQ)	<input type="checkbox"/>
Off-Hook Queue Threshold (OHQT)	0 ▾
Authcode (AUTH)	<input type="checkbox"/>

## - General Options

Input Description	Input Value
M1 is the only Controlling Party on incoming calls (CPDC)	<input type="checkbox"/>
Dial Tone on originating calls (DLTN)	<input type="checkbox"/>
Hold failure threshold (HOLD)	02 02 40
Trunk Access Restriction Group (TARG)	01
Alternate trunk route for outgoing trunks (STEP)	<input type="text"/> Range: 0 - 511
Actual outgoing toll digits to be ignored for Code Restriction (OABS)	<input type="text"/>
Display IDC Name (DNAM)	<input type="checkbox"/>
Enable Equal Access Restrictions (EOAR)	<input type="checkbox"/>
ACD DNIS route (DNIS)	<input type="checkbox"/>
Include DNIS number in CDR records (DCDR)	<input type="checkbox"/>

## Routes and Trunks Configuration continued.

## - Advanced Configurations

Input Description	Input Value
Malicious Call Trace Alarm is allowed for external calls (ALRM)	<input type="checkbox"/>
Allow last Re-directing Number (ARDN)	ARDN (NO) ▾
ANI identifier number (ANTK)	<input type="text"/>
AC 15 Timed Reminder Recall (ATRR)	<input type="checkbox"/>
Auto terminate (AUTO)	<input type="checkbox"/>
Collect Call Blocking Allowed (CCBA)	<input type="checkbox"/>
Call Forward Restriction (CFWR)	<input type="checkbox"/>
Maximum number of CNI digits (CLEN)	1 ▾
Time (in seconds) that an extension is allowed to ring or be On-hold or Call Park before the trunk is disconnected (DCTI)	0 <span style="color: green;">Range: 0 - 511</span>
North American Distinctive Ringing for incoming calls (DRNG)	<input type="checkbox"/>
Home Local Number (HLCL)	<input type="text"/>
Home National Number (HNTN)	<input type="text"/>
In-Band Automatic Number Identification route (IANI)	<input type="checkbox"/>
Incoming Identifier Send (ICIS)	<input checked="" type="checkbox"/>
Internal/external definition (IDEF)	Use network info (NET) ▾
Identify Originating Party (IDOP)	<input type="checkbox"/>
Insert (INST)	<input type="text"/>
Manual Outgoing trunk route (MANO)	<input type="checkbox"/>
Manual Route (MNL)	<input type="checkbox"/>
Music On-Hold (MUS)	<input type="checkbox"/>
North American Toll scheme (NATL)	<input checked="" type="checkbox"/>
Outgoing Identifier Send (OGIS)	<input checked="" type="checkbox"/>
Off-Hook Timer Delay (OHTD)	<input type="checkbox"/>
Outpulsing Route (OPR)	<input type="checkbox"/>
Pseudo Answer (PANS)	<input checked="" type="checkbox"/>
Periodic Clearing Signal (PECL)	<input type="checkbox"/>
Priority Level (PLEV)	2 ▾
Protocol Selection (PSEL)	DM-DM Protocol Selection (DMDM) ▾

Preference Trunk Usage Threshold (PTUT)	<input type="text" value="0"/>	Range: 0 - 510
Port Type at far end (PTYP)	Analog TIE trunks (ATT) <input type="button" value="v"/>	
Route traffic information in ACD Reports (RACD)	<input type="checkbox"/>	
Radio Paging Route (RPA)	<input type="checkbox"/>	
Route Number (RTN)	<input type="text" value=""/>	Range: 0 - 511
Satellite used for trunk route (SAT)	<input type="checkbox"/>	
Scheduled Access Restriction Group (SGRP)	<input type="text" value="0"/>	Range: 0 - 999
Special Service List number (SSL)	<input type="button" value="v"/>	
Standard Signaling Type (STYP)	Standard Data (SDAT) <input type="button" value="v"/>	
CPP/CPPO flag for incoming non-ISDN trunk call tandemed to this trunk route (TCPP)	<input type="checkbox"/>	
Tone Detector required (TDET)	<input type="checkbox"/>	
Trunk Identity (TIDY)	<input type="text" value="8011 11"/>	
Tromboning (TRMB)	<input checked="" type="checkbox"/>	
Recall signal (may not) may be received and transmitted on this route (TRRL)	<input type="checkbox"/>	
Tone Table number (TTBL)	<input type="button" value="v"/>	
Answer an Attendant Extended Call over VNS immediately on the incoming bearer trunk (VRAT)	<input type="checkbox"/>	

You can also use CLI to configure Routes and Trunks:

Use LD 14 to configure SIP Trunks, One Virtual Trunk is required for each connection. (MSNV) Media Security should be Never.

```

DES SIP
TN 100 0 02 00 VIRTUAL
TYPE IPTI
CDEN 8D
CUST 0
XTRK VTRK
ZONE 000
LDOP BOP
NMUS NO
TRK ANLG
NCOS 0
RTMB 11 1
CHID 33
TGAR 0
STRI/STRO IMM IMM
SUPN YES
AST NO
IAPG 0
CLS UNR DTN CND ECD WTA LPR APN THFD XREP SPCD MSNV (SRTP=Never)
P10 NTC MID
TKID
AACR NO
DATE 12 NOV 2006

```



Routes and Trunks Configuration continued  
Use LD 16 to configure SIP Route.

TYPE RDB  
CUST 00  
**ROUT 11**  
DES SIP  
TKTP TIE  
M911P NO  
ESN NO  
RPA NO  
CNVT NO  
SAT NO  
RCLS EXT  
VTRK YES  
ZONE 000  
PCID SIP  
CRID YES  
NODE 434  
DTRK NO  
ISDN YES  
    MODE ISLD  
    DCH 10  
    IFC SL1  
    PNI 00001  
    NCNA YES  
    NCRD YES  
    TRO YES  
    FALT NO  
    CTYP UKWN  
    INAC YES



ISAR NO  
DAPC NO  
PTYP ATT  
AUTO NO  
DNIS NO  
DCDR NO  
ICOG IAO  
SRCH LIN  
TRMB YES  
STEP  
ACOD 8011  
TCPP NO  
TARG 01  
CLEN 1  
BILN NO  
OABS  
INST  
ANTK  
SIGO STD  
STYP SDAT  
MFC NO  
ICIS YES  
OGIS YES  
PTUT 0  
TIMR ICF 1920  
OGF 1920  
EOD 13952  
DSI 34944  
NRD 10112  
DDL 70  
ODT 4096  
RGV 640  
GTO 896  
GTI 896  
SFB 3  
NBS 2048  
NBL 4096  
IENB 5  
TFD 0  
VSS 0  
VGD 6  
EESD 1024  
SST 5 0  
DTD NO  
SCDT NO  
2 DT NO  
NEDC ORG  
FEDC ORG

CPDC NO  
DLTN NO  
HOLD 02 02 40  
SEIZ 02 02  
SVFL 02 02  
DRNG NO  
CDR NO  
NATL YES  
SSL  
CFWR NO  
IDOP NO  
VRAT NO  
MUS NO  
PANS YES  
MANO NO  
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 8011 11  
ATTR NO  
TRRL NO  
SGRP 0  
CCBA NO  
ARDN NO  
ANIE 0  
CAC\_CIS 3  
AACR NO



Routes and Trunks Configuration continued  
Use LD 86 to configure (RLI) Route List Block. RLI for SIP calls.

**RLI 11**

ENTR 0

LTER NO

**ROUT 11**

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

DORG NO

SBOC NRR

IDBB DBD

IOHQ NO

OHQ NO

CBQ NO

ISET 0

NALT 5

MFRL 0

OVLL 0

### **MGC Trunk Configuration on Nortel Communication server 1000**

Use EM to configure MGC

## CS 1000 ELEMENT MANAGER

Managing: 10.10.132.134

System » IP Network » Media Gateways » IPMG 0 0 Media Gateway Controller (MGC) Configuration

### IPMG 0 0 Media Gateway Controller (MGC) Configuration

#### - Media Gateway Controller

**Hostname**  ▲  
**Management LAN (ELAN) IP address**   
**Management LAN (ELAN) gateway IP address**   
**Management LAN (ELAN) subnet mask**   
**Voice LAN (TLAN) IP address**   
**Voice LAN (TLAN) gateway IP address**   
**Voice LAN (TLAN) subnet mask**

#### - DSP Daughterboard 1

**Type of the DSP Daughterboard**  ▼  
**Voice LAN (TLAN) IP address**   
**Voice LAN (TLAN) gateway IP address**   
**Voice LAN (TLAN) subnet mask**   
**Hostname**  ▲

#### - DSP Daughterboard 2

**Type of the DSP Daughterboard**  ▼  
**Voice LAN (TLAN) IP address**   
**Voice LAN (TLAN) gateway IP address**   
**Voice LAN (TLAN) subnet mask**   
**Hostname**  ▲

#### - VGW and IP phone codec profile

**Enable Echo canceller** ☒  
**Echo canceller tail delay**  ▼  
**Voice activity detection threshold**  Range: 0 to 4  
**Idle noise level**  Range: 0 to 1  
**DTMF Tone detection** ☒  
**Enable Modem/Fax pass through mode** ☒  
**Enable V.21 FAX tone detection** ☒  
**Fax TCF Method**  Range: 1 to 2  
**FAX maximum rate (bps)**  ▼  
**FAX playout nominal delay**  Range: 0 to 300

FAX no activity timeout  Range: 10 to 32000

FAX packet size

**- Codec G711** ☒ **Select**

Codec Name G711

Voice payload size (ms/frame)

Voice playout (jitter buffer) nominal delay

Modifications may cause changes to dependent settings

Voice playout (jitter buffer) maximum delay

Modifications may cause changes to dependent settings

VAD ☐

**- Codec G729A** ☐ **Select**

Codec Name G729A

Voice payload size (ms/frame)

Voice playout (jitter buffer) nominal delay

Modifications may cause changes to dependent settings

Voice playout (jitter buffer) maximum delay

Modifications may cause changes to dependent settings

VAD ☐

**- Codec G723.1** ☐ **Select**

Codec Name G723.1

Voice payload size (ms/frame)

Voice playout (jitter buffer) nominal delay

Modifications may cause changes to dependent settings

Voice playout (jitter buffer) maximum delay

Modifications may cause changes to dependent settings

VAD ☐

**- Codec T38 FAX** ☒ **Select**

Codec Name T38 FAX

**- QoS**

Diffserv Codepoint(DSCP) Control packets  Range: 0 to 63

Diffserv Codepoint(DSCP) Voice packets  Range: 0 to 63

**- LAN configuration**

**Embedded LAN (ELAN) configuration**

Primary Call server IP address

Primary Call server Hostname

Signaling port

Broadcast port  Range: 1024 to 65535

**Telephony LAN (TLAN) configuration**

Signaling port

Voice port  Range: 1024 to 65535

**\*Mandatory fields of current configuration**

Copyright © 2002-2007 Nortel Networks. All rights reserved.



You can also use CLI to configure Routes and Trunks:

Use LD 14 to configure Trunks for MGC, One Virtual Trunk is required for each channel. 32 Virtual trunks for low density MGC daughter board and 96 for high density.

DES Low Density DB

**TN 000 0 00 01 VIRTUAL (Channel #1)**

TYPE VGW

CUST 0

XTRK DB32

ZONE 000

DES Low Density DB

**TN 000 0 00 02 VIRTUAL (Channel #2)**

TYPE VGW

CUST 0

XTRK DB32

ZONE 000

.

.

.

.

.

.

.

.

.

.

.

.

.

.

.

DES Low Density DB

**TN 000 0 00 32 VIRTUAL (Channel #32)**

TYPE VGW

CUST 0

XTRK DB32

ZONE 000

## **CDP Steering Code Configuration on Nortel Communication server 1000**

Use EM to configure CDP

**CS 1000 ELEMENT MANAGER**

Managing: **10.10.132.134**  
 Dialing and Numbering Plans » [Electronic Switched Network \(ESN\)](#) » Customer 00 » Coordinated Dialing Plan (CDP) » [Distant Steering Code List](#) » Distant Steering C

---

### Distant Steering Code

Input Description	Input Value
Distant Steering Code (DSC):	<input style="width: 100%;" type="text" value="36"/>
Flexible Length number of digits (FLEN):	<input style="width: 50%;" type="text" value="5"/> ( 0 - 10 )
Display (DSP):	<input style="width: 100%;" type="text" value="Local Steering Code (LSC)"/>
Remote Radio Paging Access (RRPA):	<input type="checkbox"/>
Route List to be accessed for trunk steering code (RLI):	<input style="width: 50%;" type="text" value="11"/>
Collect Call Blocking (CCBA):	<input type="checkbox"/>
maximum 7 digit NPA code allowed (NPA):	<input style="width: 100%;" type="text"/>
maximum 7 digit NXX code allowed (NXX):	<input style="width: 100%;" type="text"/>

You can also use CLI to configure CDP

**DSC 36 (36XXX is SIP route to Cisco Unified Communications Manager)**

FLEN 5

DSP LSC

RRPA NO

**RLI 11**

CCBA NO

NPA

NXX





\*\*\*\*\***END**\*\*\*\*\*

The information in this document is subject to change without notice. The statements, configurations, technical data, and recommendations in this document are believed to be accurate and reliable, but are presented without express or implied warranty. Users must take full responsibility for their applications of any products specified in this document.



The information in this document is proprietary to Nortel Networks. Nortel, the Nortel logo and the Globemark are trademarks of Nortel Networks.