



## **Avaya Solution & Interoperability Test Lab**

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# **Application notes for G-Tek SIP Telephone GE-202 Version 1820x.27.1.01 with Avaya™ Communication Server 1000 SIP Line Release 6.0 – Issue 1.0**

### **Abstract**

These Application Notes describe a solution comprised of Avaya™ Communication Server 1000 SIP Line Release 6.0 and the G-Tek SIP phones GE-202 firmware version 1820x.27.1.01. During the compliant testing, the GE-202 was able to register, as a SIP Client endpoint, with the Communication Server 1000. The GE-202 was able to place and receive calls from the Communication Server 1000 Release 6.0 non-SIP and SIP Line clients. The tests of other telephony features were executed such as transfer, DTMF relay, bridge line appearance, dual call, etc.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

# 1. Introduction

These application notes provide detail configurations of Avaya Communication Server 1000 SIP Line rel. 6.0 (hereafter referred to as CS1000) and SIP phone GE-202 rel. 1820X.27.1.01 (hereafter referred to as GE-202) during the compliance testing session. The GE-202 was tested against the non-SIP and SIP clients of the CS1000 SIP line 6.0. All the applicable telephony feature test cases of release 6.0 SIP line were executed on the GE-202, where applicable, to ensure the interoperability with CS1000.

## 1.1. Interoperability Compliance Testing

The focus of this compliance testing is to verify that the GE-202 is able to interoperate with our CS1000 SIP line system. The following interoperability areas were covered:

- Registration of GE-202 to the CS1000.
- Calls establishment of GE-202 with Avaya SIP and non-SIP phones on the CS1000.
- Calls establishment with emulated PSTN phones
- Telephony features: DTMF transmission, voicemail with MWI notification, busy, hold, speed dial, group call pickup, call waiting, ring again busy/no answer.
- Specific hospitality feature requirements; bridge line appearance (BLA), dual call arrangement, multiple line appearance and multiple appearances Directory Number.
- Codec negotiation

## 1.2. Support

For technical support on G-Tek SIP telephones, please contact G-Tek technical support at:

- Telephone: +886-2-26962665 ext. 221
- E-mail: support@G-Tek.com.tw

# 2. Reference Configuration

Figure 1 illustrates the test configuration used during the compliant testing event between the Avaya CS1000 and the GE202.

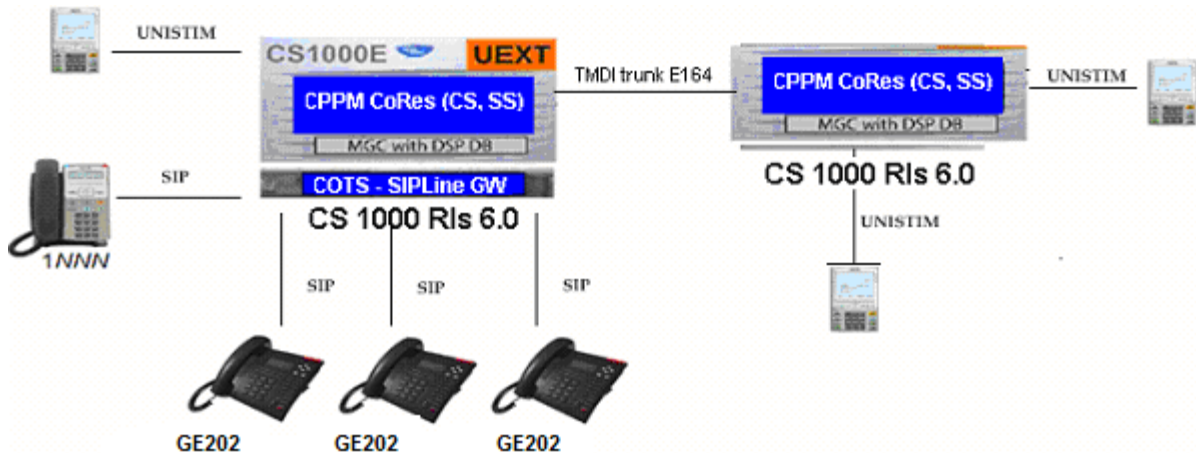


Figure 1: Lab Diagram

### 3. Equipment and Software Validated

System	Software/Loadware Version
CS1000	<ul style="list-style-type: none"> <li>Call Server (CPPM): 6.00RJ</li> <li>Signaling Server (CPPM): 6.00.18</li> <li>SIP Line Gateway (HP DL320)</li> </ul>
Call Pilot	<ul style="list-style-type: none"> <li>CallPilot (600r): 05.00.41.29</li> </ul>
11xx SIP client (Sigma)	<ul style="list-style-type: none"> <li>02.02.16.00</li> </ul>
SIP soft-phones	<ul style="list-style-type: none"> <li>SMC3456: v2.6 Build 53715</li> </ul>
IP phones	<ul style="list-style-type: none"> <li>2050PC: 3.02.0045</li> </ul>
GE202	<ul style="list-style-type: none"> <li>1820X.27.1.01</li> </ul>

### 4. Configure the Avaya CS1000 - SIP LINE

This section describes the steps to configure SIP Line using CS1000 Element Manager. A command line interface (CLI) option is available to provision the SIP Line application on CS1000 system. For detailed information, see [1]

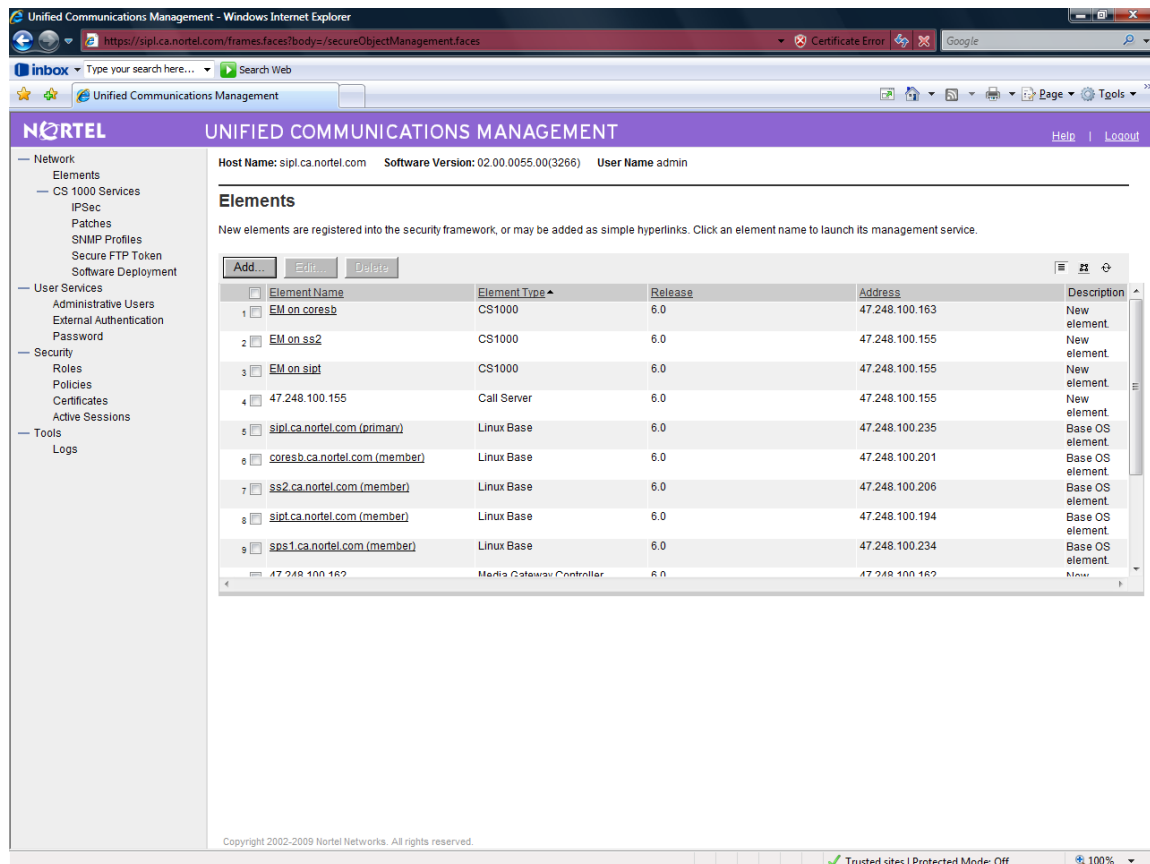
#### 4.1. Prerequisite

- CS1000 system has been upgraded to Release 6.0. For more information, see [6]
- A server which has been
  - o Installed with CS1000 Release 6.0 Linux Base.
  - o Joined CS1000 Release 6.0 Security Domain.
  - o Deployed with SIP Line Application.
 For more information, see [7].
- The following packages are enabled in the keycode.

Package Mnemonic	Package Number	Package Description	Package Type (New or Existing or Dependency)	Applicable Market
SIP_LINES	417	SIP Line Service package	New package	Global
FFC	139	Flexible Feature Codes	Existing package	Global
SIPL_NORTEL	415	Nortel SIP Line package	Existing package	--
SIPL_3RDPARTY	416	Third-Party SIP Line Package	Existing package	--

#### 4.2. Log in to Unified Communications Management (UCM) and Element Manager (EM)

- Using IE to launch CS1000 UCM web portal at <http://<IP Address or FQDN>> where <IP address or FQDN> is the UCM Framework IP address or FQDN for UCM server.
- Login with the username/password which was defined during the primary security server configuration. For more information, see [8]



**Figure 2: UCM Home Page**

- On the Elements page of Unified Communications Management, under the Element Name column, click the server name to navigate to Element Manager for that server. The CS1000 Element Manager page appears as in figure 3 below.

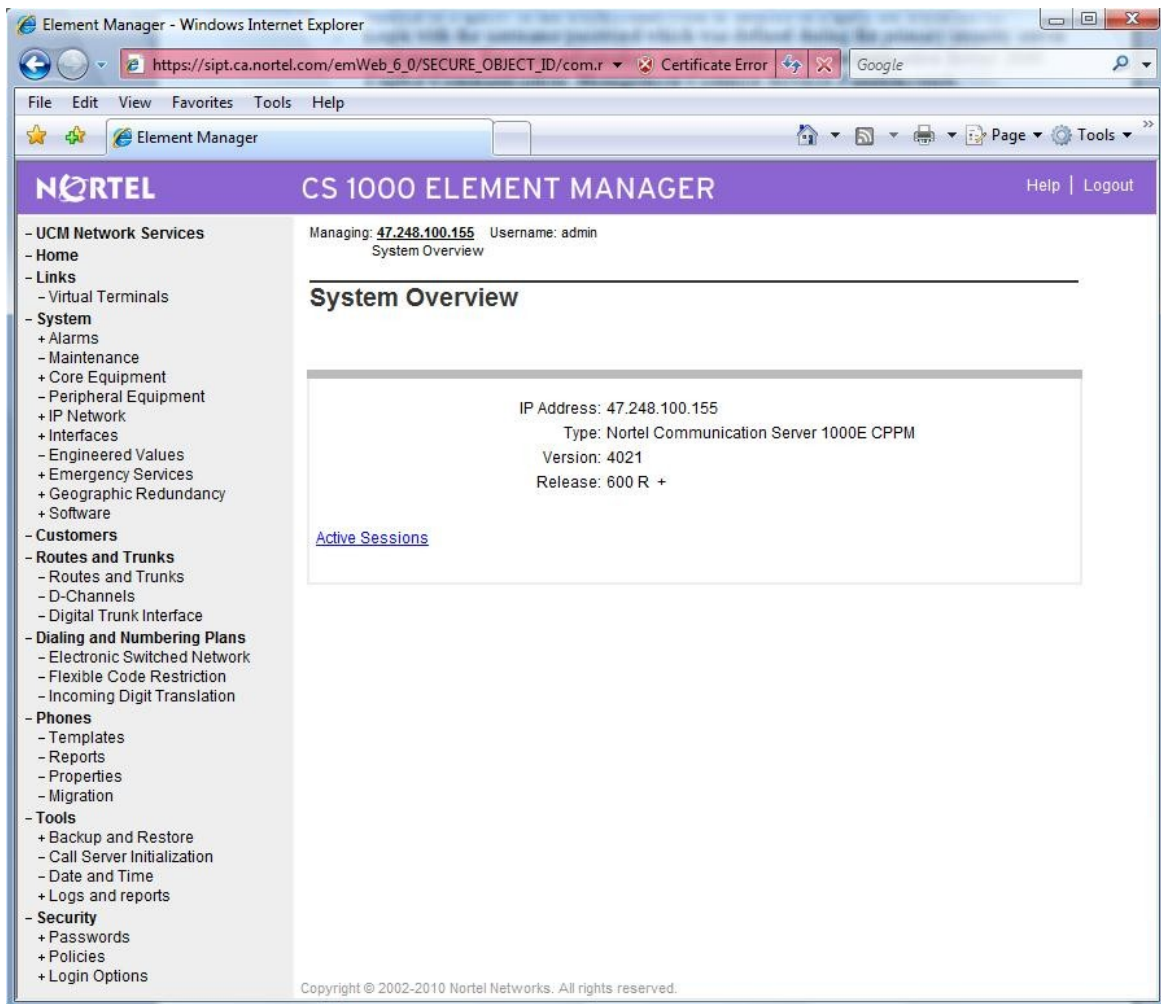
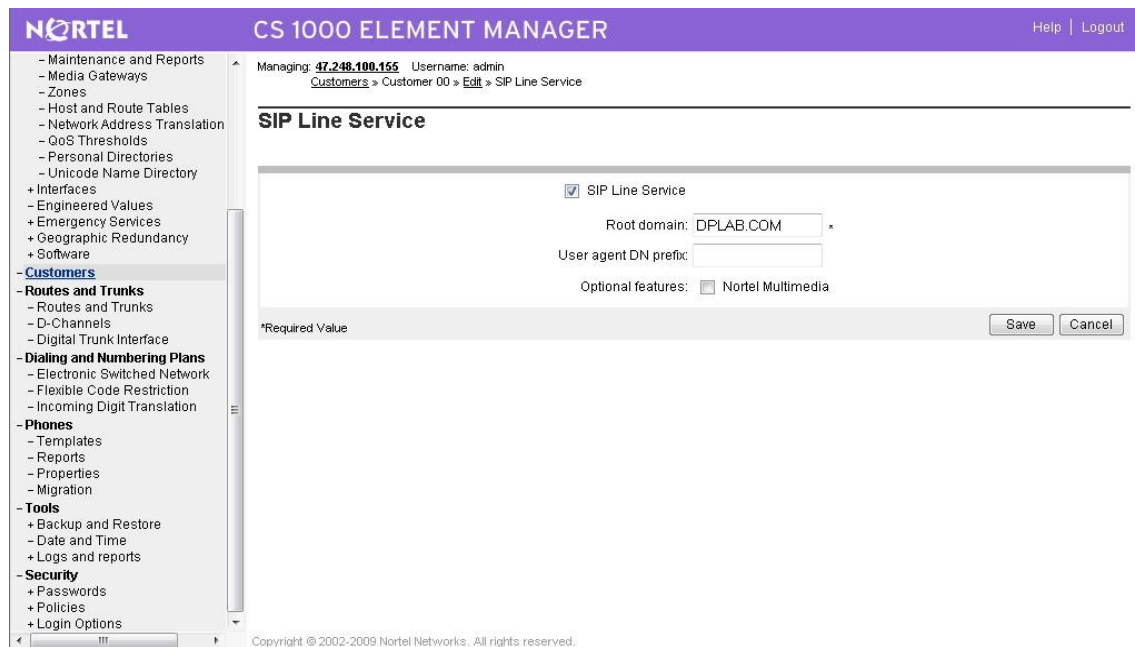


Figure 3: CS1000 EM Home Page

#### 4.3. Enable SIP Line Service and Configure the Root Domain in Customer Data Block (CDB)

- On the EM page, navigate to **Customers**, select the customer number to be enabled with SIP Line Service.
- Enable SIP Line Service by clicking on the **SIP Line Service** check box.
- Enter the SIP Line **Root Domain** name in the **Root Domain** text box.



**Figure 4: SIP Line Service in Customers Data Block**

#### 4.4. SIP Line Telephony Node Configuration

- On the EM page, navigate to **System** → **IP Network** → **Nodes: Servers, Media Cards**.
- Click **Add** to add a new SIP Line Node to IP Telephony Nodes. To see the SIP Line node details, click on the SIP Line Node ID.
- Enter Node ID in the **Node ID** text box.
- Enter Call Server IP Address in the **Call Server IP Address** text box.
- Enter Node IP Address in the **Node IP Address** text box.
- Enter TLAN Subnet Mask in the **Subnet Mask** text box.
- Enter ELAN Gateway IP Address in the **Gateway IP Address** text box.
- Enter ELAN Subnet Mask in the **Subnet Mask** text box.
- Check **SIP Line** check box to enable SIP Line for this Node.

**NORTEL CS 1000 ELEMENT MANAGER** Help | Logout

Managing: 47.248.100.155 Username: admin  
System » IP Network » IP Telephony Nodes

### New IP Telephony Node

Step 1: Define the new Node and its services.  
You will also require pre-configured servers with appropriate application software already deployed to host the selected services.

Node ID: 556 \* (0-9999)

Call Server IP Address: 47.248.100.155 \*

**Telephony LAN (TLAN)** **Embedded LAN (ELAN)**

Node IP Address: 47.248.100.237 \* Gateway IP address: 47.248.100.129 \*

Subnet Mask: 255.255.255.240 \* Subnet Mask: 255.255.255.224 \*

Applications ☒ SIP Line

☐ UNISim Line Terminal Proxy Server (LTPS)

☐ Virtual Trunk Gateway (SIPGw, H323Gw)

☐ Personal Directory (PD)

☐ Presence Publisher

\* Required Value. Next > Cancel

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**Figure 5 – IP Telephony Node**

- Click **Next**. The page to add server to node appears (not shown).
- On Add Server page, from the **Please Select Server** list, select the server to add to the node.
- Click **Add** (Do not click the Next button).
- Select the check box next to the newly added server, and click **Make Leader**.
- Figure 6 shows server “sipl” selected

**NORTEL CS 1000 ELEMENT MANAGER** Help | Logout

Managing: 47.248.100.155 Username: admin  
System » IP Network » IP Telephony Nodes

### New IP Telephony Node (ID:557)

Step 2: Associate required signaling servers for SIP Line services.  
In order to appear in the list below, servers must already be defined within ECM, should not be part of any other IP telephony node and deployed application(s) on the server(s) should match the service(s) selected for this node.

sipl Add Remove Make Leader Print Refresh

<input type="checkbox"/>	Hostname	Type	Deployed Applications	ELAN IP	TLAN IP	Role
<input checked="" type="checkbox"/>	sipl					

Select from the list above and click Add to associate servers with this node.  
Selected servers must have identical application deployments.

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**Figure 6 – IP Telephony Node – Add Server**



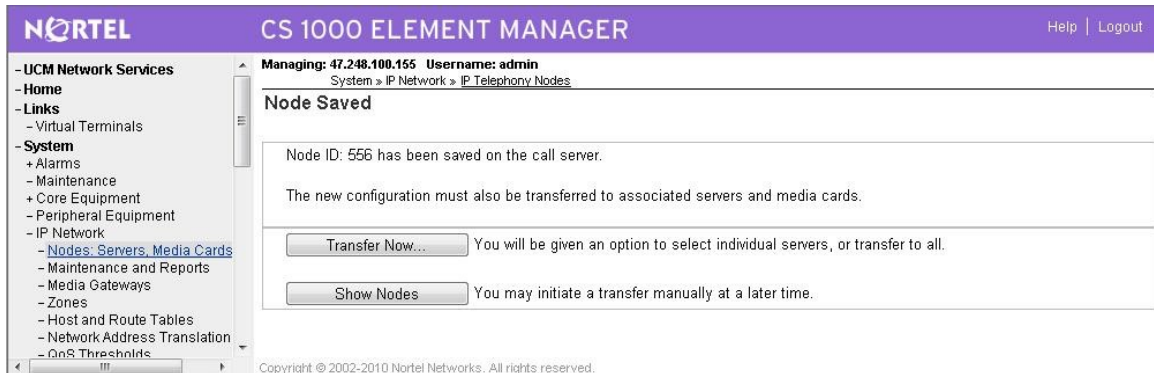
- Click **Next**. The SIP Line Configuration Detail page appears.
- Enter SIP Line domain name in **SIP Domain name** text box. This must be the same as the domain name configured in **Customers**, section 4.3.

**Figure 7 – SIP Line Node Details**

- Under the **SIP Line Gateway Services** section, select **MO** from the **SLG Role** list.
- From the **SLG Mode** list, select **S1/S2** (SIP Proxy Server 1 and Server 2).

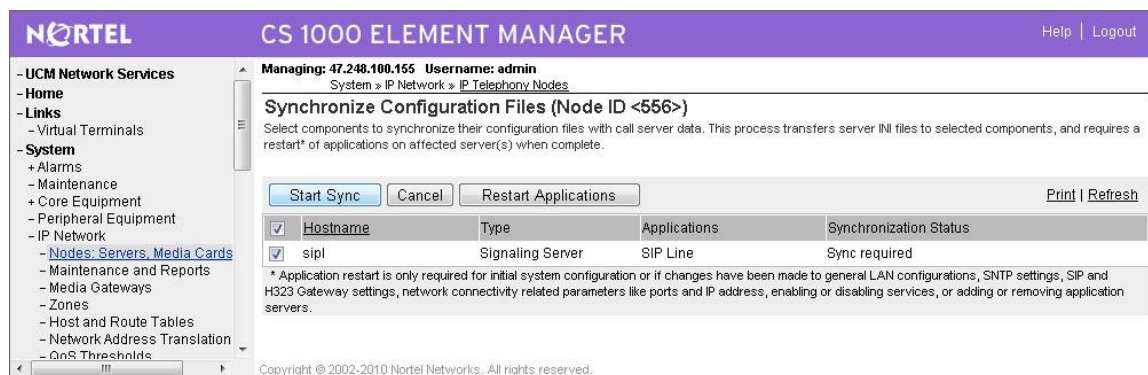
**Figure 8 – SIP Line Node Details (cont.)**

- Click **Next**. The **Confirm new Node details** page appears.
- Click **Finish** and wait for the configuration to be saved. The **Node Saved** page then appears.



**Figure 9 – Transfer Configuration**

- Click **Transfer Now**. The **Synchronize Configuration Files (Node ID 556)** page appears.
- Select some or all of the node elements and then click **Start Sync** to transfer the configuration files to the selected servers.



**Figure 10 – Synchronize Configuration Files**

## 4.5. D-Channel over IP Configuration

- On the EM page, navigate to **Routes and Trunks** → **D-Channels**.
- Under the **Configuration** section, from the **Choose a D-Channel Number** list, select a D-Channel number (not shown).
- Under the Configuration section, from the **Type** list, select **DCH**.
- Click to **Add**.
- From the **D channel Card Type (CTYP)** list, select **D-Channels is over IP (DCIP)**.
- Click to **Add**.
- The **D-Channels xx Property Configuration** page appears, as shown in Figure 11.
- From the **Interface type for D-channel (IFC)** list, select **Meridian Meridian1 (SL1)**.

- Click the **Basic options (BSCOPT)** link. The **Basic options (BSCOPT)** list expands.
- Click **Edit** to configure **Remote Capabilities (RCAP)**.

Managing: 47.248.100.155 Username: admin  
Routes and Trunks > D-Channels > D-Channels 30 Property Configuration

### D-Channels 30 Property Configuration

**- Basic Configuration**

Input Description	Input Value
Action Device And Number (ADAN) (TYPE)	DCH
D channel Card Type (CTYP)	DCIP
Designator (DES)	SIPLine
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)	5
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
<b>- Basic options (BSCOPT)</b>	
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)	<a href="#">Edit</a>
+ - Change protocol timer value (TIMR)	
- B channel Service messaging. (BSRV)	<input type="checkbox"/>
+ Advanced options (ADVOPT)	
+ Feature Packages	

Submit Refresh Delete Cancel

Figure 11 – SIP Line D-Channel Property Configuration

- Figure 12 now appears.
- Select the **Message waiting interworking with DMS-100 (MWI)** check box. This must be enabled to support voice mail notification on SIP Line endpoints.
- Select the **Network name display method 2 (ND2)** check box. This must be enabled to support name display between SIP Line endpoints.
- At the bottom of the **Remote Capabilities Configuration** page, click **Return - Remote Capabilities**.
- The **D-Channel xx Property Configuration** page reappears. Click **Submit** button shown at the bottom of the page.

**CS 1000 ELEMENT MANAGER**
[Help](#) | [Logout](#)

- UCM Network Services
- Home
- Links
  - Virtual Terminals
- System
  - + Alarms
  - Maintenance
  - + Core Equipment
  - Peripheral Equipment
  - IP Network
    - Nodes: Servers, Media Cards
    - Maintenance and Reports
    - Media Gateways
    - Zones
    - Host and Route Tables
    - Network Address Translation
    - QoS Thresholds
    - Personal Directories
    - Unicode Name Directory
  - + Interfaces
  - Engineered Values
  - Emergency Services
  - + Geographic Redundancy
  - + Software
- Customers
  - Routes and Trunks
    - Routes and Trunks
    - **D-Channels**
      - Digital Trunk Interface
- Dialing and Numbering Plans
  - Electronic Switched Network
  - Flexible Code Restriction
  - Incoming Digit Translation
- Phones
  - Templates
  - Reports
  - Properties
  - Migration
- Tools
  - + Backup and Restore
  - Date and Time
  - + Logs and reports
- Security
  - + Passwords
  - + Policies
  - + Login Options

Managing: **47.248.100.155** Username: admin  
Routes and Trunks » [D-Channels](#) » [D-Channels 30 Property Configuration](#) » - Remote Capabilities Configuration

### - Remote Capabilities Configuration

Input Description	Input Value
Basic rate interface (BRI)	<input type="checkbox"/>
Call completion on busy using integer value (CCBI)	<input type="checkbox"/>
Call completion on busy using object identifier (CCBO)	<input type="checkbox"/>
Call completion on busy for QSIG and EuroISDN BRI (CCBS)	<input type="checkbox"/>
Call completion on no response using integer value (CCNI)	<input type="checkbox"/>
Call completion on no response using object identifier (CCNO)	<input type="checkbox"/>
Call completion to no reply for QSIG and EuroISDN BRI (CCNR)	<input type="checkbox"/>
Network call park (CPK)	<input type="checkbox"/>
Connected line identification presentation (COLP)	<input type="checkbox"/>
Call transfer integer (CTI)	<input type="checkbox"/>
Call transfer object (CTO)	<input type="checkbox"/>
Diversion info. is sent using integer value (DV1)	<input type="checkbox"/>
Diversion info. is sent using object identifier (DV1O)	<input type="checkbox"/>
Rerouting requests processed using integer value (DV2)	<input type="checkbox"/>
Rerouting requests processed using object identifier (DV2O)	<input type="checkbox"/>
Diversion info. sent. rerouting requests processed (DV3)	<input type="checkbox"/>
EuroISDN - div. info sent. rerouting req. processed (DV3O)	<input type="checkbox"/>
Call transfer notification and invocation to EuroISDN (ECTO)	<input type="checkbox"/>
Malicious call identification (MCID)	<input type="checkbox"/>
MCDN QSIG conversion (MQC)	<input type="checkbox"/>
Remote D-channel is on a MSDL card (MSL)	<input type="checkbox"/>
Message waiting interworking with DMS-100 (MWM)	<input checked="" type="checkbox"/>
Network access data (NAC)	<input type="checkbox"/>
Network call trace supported (NCT)	<input type="checkbox"/>
Network name display method 1 (ND1)	<input type="checkbox"/>
Network name display method 2 (ND2)	<input checked="" type="checkbox"/>
Network name display method 3 (ND3)	<input type="checkbox"/>
Name display - integer ID coding (NDI)	<input type="checkbox"/>
Name display - object ID coding (NDO)	<input type="checkbox"/>
Path replacement uses integer values (PRI)	<input type="checkbox"/>
Path replacement uses object identifier (PRO)	<input type="checkbox"/>
Release Link Trunks over IP (RLTI)	<input type="checkbox"/>
Remote virtual queuing (RVQ)	<input type="checkbox"/>
Trunk anti-tromboning operation (TAT)	<input type="checkbox"/>
User to user service 1 (UUS1)	<input type="checkbox"/>
NI-2 name display option. (NDS)	<input type="checkbox"/>
Message waiting indication using integer values (QMWI)	<input type="checkbox"/>
Message waiting indication using object identifier (QMWVO)	<input type="checkbox"/>
User to user signalling (UUI)	<input type="checkbox"/>

**Figure 12 – SIP Line D-Channel RCAP Configuration Details**

QT; Reviewed:  
SPOC 4/9/2010

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12 of 31  
GE-202\_CS1000



## 4.6. Application Module Link (AML) over Embedded LAN (ELAN) Configuration

- On the EM page, navigate to *System* → *Interfaces* → *Application Module Link*.
- Click *Add* to add an Application Module Link. *New Application Module Link* page appears.
- Enter AML port in the *Port number* text box. The SIP Line Service can use ports 32 to port 127. In this case, SIP Line Service is configured to use port 32.
- Click *Save* to save the configuration.

The screenshot shows the 'New Application Module Link' configuration page in the Nortel CS 1000 Element Manager. The page has a purple header with the Nortel logo and 'CS 1000 ELEMENT MANAGER'. On the left is a navigation tree with categories like 'UCM Network Services', 'Home', 'Links', 'System', 'Customers', and 'Routes and Trunks'. The 'Application Module Link' option under 'System' is selected. The main content area shows the configuration for a new AML link. It includes a 'Port number' field set to 32, a 'Description' field set to 'SIPLine', and a 'Maximum octets' dropdown set to 512. There is a checkbox for 'Link control system parameters' which is unchecked. At the bottom right are 'Save' and 'Cancel' buttons. The footer contains the copyright notice: 'Copyright © 2002-2010 Nortel Networks. All rights reserved.'

Figure 13 – Application Module Link Configuration

## 4.7. Value Added Server (VAS) Configuration

- On the EM page, navigate to *System* → *Interfaces* → *Value Added Server*.
- Click *Add* to add new Value Added Server. The *Add Value Added Server* page appears.
- Click on the *Ethernet LAN Link*.
- Enter the Ethernet LAN Link number in the *Ethernet LAN Link* text box.
- Ensure that the *Application Security* check box is unchecked.

**NORTEL CS 1000 ELEMENT MANAGER** Help | Logout

Managing: 47.248.100.155 Username: admin  
System > Interfaces > Value Added Server > Edit Value Added Server 032

### Edit Value Added Server 032

Ethernet LAN Link: 032  
ELAN port configured in ADAN

Application Security: ☒   
Interval: 1   
Time interval for checking the link for overload in five second increments

Message Count Threshold: 9999 \* (10 - 9999)

Save Cancel

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Figure 14 – Value Added Service for Application Module Link

## 4.8. Virtual Trunk Zone Configuration

- On the EM page, navigate to *System* → *IP Network* → *Zones*.
- On the *Zones* page, select *Bandwidth Zones*.
- On the *Bandwidth Zones* page, select a *Bandwidth Zone number* from the list, and click to *Add*.
- On the *Zone Basic Property and Bandwidth Management* page, set the zone properties based on bandwidth availability. It is recommended to set the *Zone Strategy* to *BestQuality (BQ)*.
- From the *Zone Intent (ZBRN)* list, select *VTRK (VTRK)*.
- Click *Submit*.

**NORTEL CS 1000 ELEMENT MANAGER** Help | Logout

Managing: 47.248.100.155 Username: admin  
System > IP Network > Zones > Bandwidth Zones > Bandwidth Zones 254 > Zone Basic Property and Bandwidth Management

### Zone Basic Property and Bandwidth Management

Input Description	Input Value
Zone Number (ZONE):	254
Intrazone Bandwidth (INTRA_BW):	100000
Intrazone Strategy (INTRA_STGY):	Best Quality (BQ)
Interzone Bandwidth (INTER_BW):	100000
Interzone Strategy (INTER_STGY):	Best Quality (BQ)
Resource Type (RES_TYPE):	Shared (SHARED)
Zone Intent (ZBRN):	VTRK (VTRK)
Description (ZDES):	

Submit Refresh Delete Cancel

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Figure 15 – Virtual Trunk Zone Configuration

## 4.9. SIP Line Route Data Block (RDB) Configuration

- On the EM page, navigate to *Routes and Trunks* → *Routes and Trunks*.
- Click *Add* to select a customer number.
- On the *Customer xx, New Route Configuration* page, from the *Route number (ROUT)* list, select a route number.
- From the *Trunk type (TKTP)* list, select *TIE trunk data block (TIE)*.
- When Trunk Type (TKTP) is selected, the following options appear:
  - Trunk type M911P (M911P)
  - The route is for a virtual trunk route (VTRK)
  - Digital trunk route (DTRK)
  - Integrated services digital network option (ISDN)
- From the *Incoming and outgoing trunk (ICOG)* field, select *Incoming and Outgoing (IAO)*.
- In the *Access code for the trunk route (ACOD)* field, enter the access code.
- Select *The route is for virtual trunk route (VTRK)* check box.
- In the *Zone for codec selection and bandwidth management (ZONE)* field, enter the zone number. (Use the same zone as configured in 4.8 “*Virtual Trunk Zone Configuration*”)
- In the *Node ID of signaling server of this route (NODE)* field, enter the node ID of the SIP Line Gateway.
- From the *Protocol ID for the route (PCID)* list, select *SIP Line (SIPL)*.
- Select the *Integrated services digital network option (ISDN)* check box.
- From the *Mode of operation (MODE)* list, select *Route uses ISDN Signaling Link (ISLD)*.
- In the *D channel number (DCH)* field, enter the D-channel number.
- From the *Interface type for route (IFC)* list, select *Meridian M1 (SL1)*.
- Ensure the *Network calling name allowed (NCNA)* and *Insert ESN Access Code (INAC)* check boxes are selected.
- For the Basic *Route Options*, *Network Options*, *General Options*, and *Advanced Configurations* sections. The default values were used.
- Click *Save*.

**NORTEL** CS 1000 ELEMENT MANAGER Help | Logout

Managing: **47.248.100.155** Username: admin  
Routes and Trunks > Routes and Trunks > Customer 0, Route 30 Property Configuration

### Customer 0, Route 30 Property Configuration

**- Basic Configuration**

Route data block (RDB) (TYPE)   
 Customer number (CUST)   
 Route number (ROUT)   
 Designator field for trunk (DES)   
 Trunk type (TKTP)   
 Incoming and outgoing trunk (ICOG) Incoming and Outgoing (IAO)   
 Access code for the trunk route (ACOD)  Range: 0 - 255  
 Trunk type M911P (M911P) ☐  
 The route is for a virtual trunk route (VTRK) ☒  
 Zone for codec selection and bandwidth management (ZONE)  Range: 0 - 255  
 Node ID of signaling server of this route (NODE)  Range: 0 - 9999  
 Protocol ID for the route (PCID)   
 Integrated services digital network option (ISDN) ☒  
 Mode of operation (MODE)   
 D channel number (DCH)  Range: 0 - 254  
 Interface type for route (IFC)   
 Private network identifier (PNI)  Range: 0 - 32700  
 Network calling name allowed (NCNA) ☒  
 Network call redirection (NCRD) ☐  
 Trunk route optimization (TRO) ☐  
 Recognition of DTI2 ABCD FALT signal for ISL (FALT) ☐  
 Channel type (CHTY)   
 Call type for outgoing direct dialed TIE route (CTYP)   
 Insert ESN access code (INAC) ☒  
 Integrated service access route (ISAR) ☐  
 Display of access prefix on CLID (DAPC) ☐  
 Mobile extension route (MBXR) ☐

**+ Basic Route Options**  
**+ Network Options**  
**+ General Options**  
**+ Advanced Configurations**

**Figure 16 –SIP Line Route Configuration**

**Note:** There is an outstanding issue (Q02073088) with the CS1000 Call Waiting feature which occurs when *Network Call Redirection* is enabled. If the Network Call Redirection feature is not required, uncheck the feature to make the Call Waiting work.

## 4.10. SIP Line Virtual Trunk Configuration

- On the EM page, navigate to *Routes and Trunks* → *Routes and Trunks*.
- Select the customer for which you are configuring Virtual Trunks.
- Click *Add trunk associated with the route listing* to add new trunk members.
- The *Customer xx, Route yy, New Trunk Configuration* Web page appears.
- Choose *Multiple trunk input number (MTINPUT)* if you are using more than one trunk.
- From the *Trunk data block (TYPE)* list, select *IP Trunk (IPTI)*.
- In the *Terminal Number (TN)* field, enter a TN.
- Enter a *Route number, Member number (RTMB)*.
- Enter a *Trunk Group Access Restriction (TGAR)* value.



- In the **Channel ID for this trunk (CHID)** field, enter a **channel ID** (where the range is 1 to 382).
- To specify a **Class of Service (CLS)** for the trunk, click **Edit**. The **Class of Service Configuration** Web page appears (not shown).
- Select a **Class of Service**.
- Click **Return Class of Service** to return to the **New Trunk Configuration** Web page.
- Select **Basic Configuration**. The **Basic Configuration** list expands.
- From the **Start arrangement Incoming (STRI)** list, select a value for the start arrangement for incoming calls.
- From the **Start arrangement Outgoing (STRO)** list, select a value for the start arrangement for outgoing calls.
- Select **Advanced Trunk Configurations**. The **Advanced Trunk Configurations** list expands (not shown).
- Configure **Network Class of Service group (NCOS)**.
- Click **Save**.

**NORTEL** CS 1000 ELEMENT MANAGER Help | Logout

**Customer 0, Route 30, Trunk 1 Property Configuration**

**- Basic Configuration**

Input Description	Input Value
Trunk data block (TYPE)	IPT1
Terminal Number (TN)	100 0 01 00
Designator field for trunk (DES)	SIPLINE
Extended Trunk (XTRK)	VTRK
Route number, Member number (RTMB)	30 1
Level 3 Signaling (SIGL)	
Card Density (CDEn)	8D
Start arrangement Incoming (STRI)	Immediate (IMM)
Start arrangement Outgoing (STRO)	Immediate (IMM)
Trunk Group Access Restriction (TGAR)	0
Channel ID for this trunk (CHID)	1
Increase or decrease the member numbers (INC)	Increase channel and member number (YES)
Class of Service (CLS)	Edit

**+ Advanced Trunk Configurations**

Save Delete Cancel

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**Figure 17 –SIP Line Trunk Configuration**

## 4.11. SIP Line Phones Configuration

Following is a sample configuration for a Third Party SIP Line endpoint. Depending on supported features and service access level of the user, this configuration can be adjusted accordingly.

>LD 11

REQ: prt

TYPE: tnb  
TN 96 0 1 27  
DATE  
PAGE  
DES

DES POLYCO  
TN 096 0 01 27 VIRTUAL  
TYPE **UEXT**  
CDEN 8D  
CTYP XDLC  
CUST 0  
UXTY **SIPL**  
MCCL **YES**  
SIPN 0 ← Set this to 1 and set SIP3 to 0 if this TN is reserved for Nortel SIP Phones  
SIP3 **1** ← Set this to 1 and set SIPN to 0 if this TN is reserved for third party SIP Phones  
FMCL 0  
TLSV 0  
SIPU **55573**  
NDID **556**  
SUPR NO  
SUBR DFLT MWI RGA CWI MSB  
UXID  
NUID  
NHTN  
CFG\_ZONE **001**  
CUR\_ZONE 001  
ERL  
ECL 0  
FDN **55576** ← If CLS FNA is equipped, call will be forwarded no answer to this number  
TGAR 0  
LDN NO  
NCOS 0  
SGRP 0  
RNPG **2** ← This field must be set first if call pickup is equipped (CLS PUA)  
SCI 0  
SSU  
XLST  
SCPW **1234**  
SFLT NO  
CAC\_MFC 0  
CLS\_UNR **FBA** WTA LPR **PUA** MTD **FNA HTA** TDD HFA CRPD  
**MWA** LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1  
POD DSX VMD SLKD CCSD **SWD** LND **CNDA**  
CFTD SFD MRD DDV **CNIA** CDCA MSID DAPA BFED RCBD  
ICDD CDMD LLCN MCTD CLBD AUTU

GPU A DPU A **DNDA** CFX A ARHD CLTD ASCD  
 CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD  
 UDI RCC HBTD AHA IPND **DDGA NAMA** MIND PRSD NRWD NRCD NROD  
 DRDD EXR0  
 USMD USRD ULAD CCBD RTDD RBDD RBHD PGND FLXD FTTC DNDY DNO3  
 MCBN  
 FDSD NOVD VOLA VOUD CDMR ICRD MCDD T87D MSNV FRA PKCH  
 CPND\_LANG ENG  
 RCO 0  
 HUNT **55576** ← If CLS HTA/FBA is equipped, call will be forwarded busy to this number  
 LHK 0  
 PLEV 02  
 DANI NO  
 AST  
 IAPG 0  
 AACS NO  
 ITNA NO  
 DGRP  
 MLWU\_LANG 0  
 MLNG ENG  
 DNDR 0  
 KEY 00 **SCR 55573** 0 MARP  
 CPND  
 CPND\_LANG ROMAN  
 NAME **Polycom 55573**  
 XPLN 13  
 DISPLAY\_FMT FIRST, LAST  
 01 HOT U **2655573** MARP 0  
 02 SCU **0004** ← Speed Call User  
 03  
 04 **MSB** ← This key can be different than key 04 to enable Make Set Busy (MBS) feature  
 05  
 06  
 07  
 08  
 09  
 10  
 11  
 12  
 13  
 14  
 15  
 16  
 17 TRN  
 18 AO6  
 19 CFW 16 55574

20 RGA  
21 PRK  
22 RNP  
23  
24 PRS  
25 CHG  
26 CPN  
27  
28  
29  
30  
31

## 4.12. PSTN Outside Trunk Configuration

*Following is a sample configuration which was used during the compliance test. For more information about PRI Trunk Configuration, see [3].*

### ***Procedure summary***

This procedure is applied for both CS1000 system under test and CS1000 PSTN simulator.

No.	Overlay	Action
1	LD 17	Adding a PRI card
2	LD 17	Adding a PRI D-Channel
3	LD 15	Defining a PRI customer
4	LD 16	Defining a PRI service route
5	LD 14	Defining service channels and PRI trunks
6	LD 73	Defining system timers and clock controller
7	LD 48	Enable TMDI or PRI MSDL card
8	LD 60	Enable Clock Controller
9	LD 60	Enable Digital trunk loop
10	LD 96	Enable D-channel

### Adding a PRI card

*The programming example below shows how to add a PRI card using LD 17. For all other fields not listed in the example press RETURN to use default values.*

Prompt	Response	Description
REQ	CHG	Change data.
TYPE	CFN	Configuration data block.
CEQU	YES	Changes to common equipment.
DLOP	10	Digital Trunk Interface Loop
MG_CARD	4 0 1	MG card assigned to superloop
MODE	PRI	Mode of operation
TMDI	YES	Card is TMDI card
TRSH	0	Threshold

### Adding a PRI D-channel

*The programming example below shows how to add a PRI D-channel using LD 17. For all other fields not listed in the example press RETURN to use default values.*

Prompt	Response	Description
REQ	CHG	Change existing data
TYPE	CFN	Configuration data block.
ADAN	NEW DCH 10	Add a primary D-channel (any unused SDI port.)  xx = 1-9 for Option 11C main cabinet, 11-19 for IP expansion cabinet 1, 21-29 for IP expansion cabinet 2, 31-39 for IP expansion cabinet 3, and 41-49 for IP expansion cabinet 4.  Xx = 11-14, 21-24, 31-34, 41-44 of the first, second, third and fourth Media Gateway, respectively.

CTYP	TMDI	Card type where:  MSDL = The NTB51BA Downloadable D-Channel Daughterboard.  TMDI = TMDI (NTRB21) card.
DES	T1_QSIG	Designator field.
USR	PRI	D-channel is for ISDN PRI only. <b>Note:</b> 2.0 Mb only supports PRI or SHA user
IFC	ISGF	Interface type.
DCHL	10	PRI card number carries the D-channel. Must match entry made for the "CDNO" associated with the "DCHI" prompt above.  Where: xx = 1-9 for Option 11C main cabinet, 11-19 for IP expansion cabinet 1, 21-29 for IP expansion cabinet 2, 31-39 for IP expansion cabinet 3, and 41-49 for IP expansion cabinet 4.  xx = 11-14, 21-24, 31-34, 41-44 of the first, second, third and fourth Media Gateway, respectively.
SIDE	NET	NET = network, the controlling switch (applied for CS1000 PSTN simulator USR = slave to the controller (applied for CS1000 system under test)
RLS	6	Software release of far-end. This is the current software release of the far-end. If the far-end has an incompatible release of software, it prevents the sending of application messages, for example, 'Network Ring Again.
RCAP	CCBI CCNI PRI DV3I CTI QMWI	Remote Capabilities.
PR_TRIGS	DIV 2 3	Path Replacement Triggers
PR_TRIGS	CNG 2 3	
PR_TRIGS	CON 2 3	
PR_TRIGS	CTR2 2 3	

### Defining a PRI customer

*The programming example below shows how to define a PRI customer using LD 15. For all other fields not listed in the example press RETURN to use default values.*

Prompt	Response	Description
REQ	CHG	Change existing data.
TYPE	CDB	Customer data block.
CUST	0	Customer number.
ISDN	YES	Customer is equipped with ISDN.

### **Defining a PRI service route**

*The programming example below shows how to add a PRI service route using LD 16. For all other fields not listed in the example press RETURN to use default values.*

Prompt	Response	Description
REQ	NEW	Create new data
TYPE	RDB	Route data block
CUST	0	Customer number
ROUT	10	Route number
DES	T1_QSIG	Designator field for trunk
TKTP	TIE	Trunk type
DTRK	YES	Digital trunk route
ISDN	YES	ISDN option
MODE	PRI	Route used for PRI only
PNI	1	Customer private network identifier. Is the same as the CDB PNI at far-end.
IFC	ISGF	Interface type.
ICOG	IAO	Incoming and outgoing
ACOD	8010	Trunk access code

### Defining service channels and PRI trunks

*The programming example below shows how to create service channels and PRI trunks using LD 14. For all other fields not listed in the example press RETURN to use default values.*

Prompt	Response	Description
REQ	NEW 23	Create 23 new trunks
TYPE	Tie	Trunk type
TN	10 1	Loop (card) and channel number for digital trunks
PCML	MU	System PCM law.
DES	T1_QSIG	Designator field for trunk
CUST	0	Customer number
RTMB	10 1	Service route number and trunk member number
CLS	UNR DTN	Trunk Class Of Service

### Defining system timers and clock controller parameters

*Note: This step is only applied for the CS1000 PSTN simulator system which keeps the clock controller.*

*The programming example below shows how to define system timers and clock controller parameters using LD 73. For all other fields not listed in the example press RETURN to use default values.*

Prompt	Response	Description
REQ	CHG	Change data.
TYPE	DDB	Digital Data Block
MGCLK	4 0 1	Card slot number for Media Gateway 4 0
PREF	1	Card number of PRI/DTI/SILC or DTI2/PRI2/SILC containing the primary clock reference.
SREF	1	Card number of PRI/DTI/SILC or DTI2/PRI2/SILC containing the primary clock reference.

### Enabling T1 QSIG Service

#### Enable TMDI card

*The example below shows how to enable TMDI card using LD 48.*



```
>ld 48
LNK000
.enl tmdi 4 0 1
```

OK

### **Enable Clock Controller**

*The example below shows how to enable clock controller using LD 60.*

```
>ld 60
DTI000
.enl cc 4 0
.OK
```

### **Enable PRI loop**

*The example below shows how to enable PRI loop using LD 60.*

```
>ld 60
DTI000
.enll 10
```

OK

.

### **Enable D-Channel**

*The D-Channel may not automatically come up. The example below shows how to enable PRI D-channel using LD 96.*

```
>ld 96
DCH000
.enl dch 10
```

.

DCH: 10 EST CONFIRM TIME: 19:38:44 30/09/2009


DCH 10 UIPE\_OMSG CC\_RESTART\_REQ REF 00000000 CH 0 TOD 19:38:44 CK  
E0DAF978  
TYPE: ALL CHANNEL

DCH 10 UIPE\_IMSG CC\_RESTART\_CONF REF 00008000 TOD 19:38:44 CK E0DAF9C2  
TYPE: ALL CHANNEL

## 5. Configure G-Tek GE202 SIP Phone

### 5.1. SIP Account Settings

Figure 18 shows the account settings used for the SIP phone.



The image shows a web configuration interface for a G-Tek GE202 SIP phone. The header features a phone icon and the text "Web Configuration" with a red double arrow. A left sidebar contains a navigation menu with the following items: Phone Settings, System Settings, Global SIP Settings, SIP Accounts, and Network. The main content area is titled "SIP Account Settings" and includes a sub-header "SIP Account 1". Below this, a form contains various settings for the SIP account. The settings are as follows:

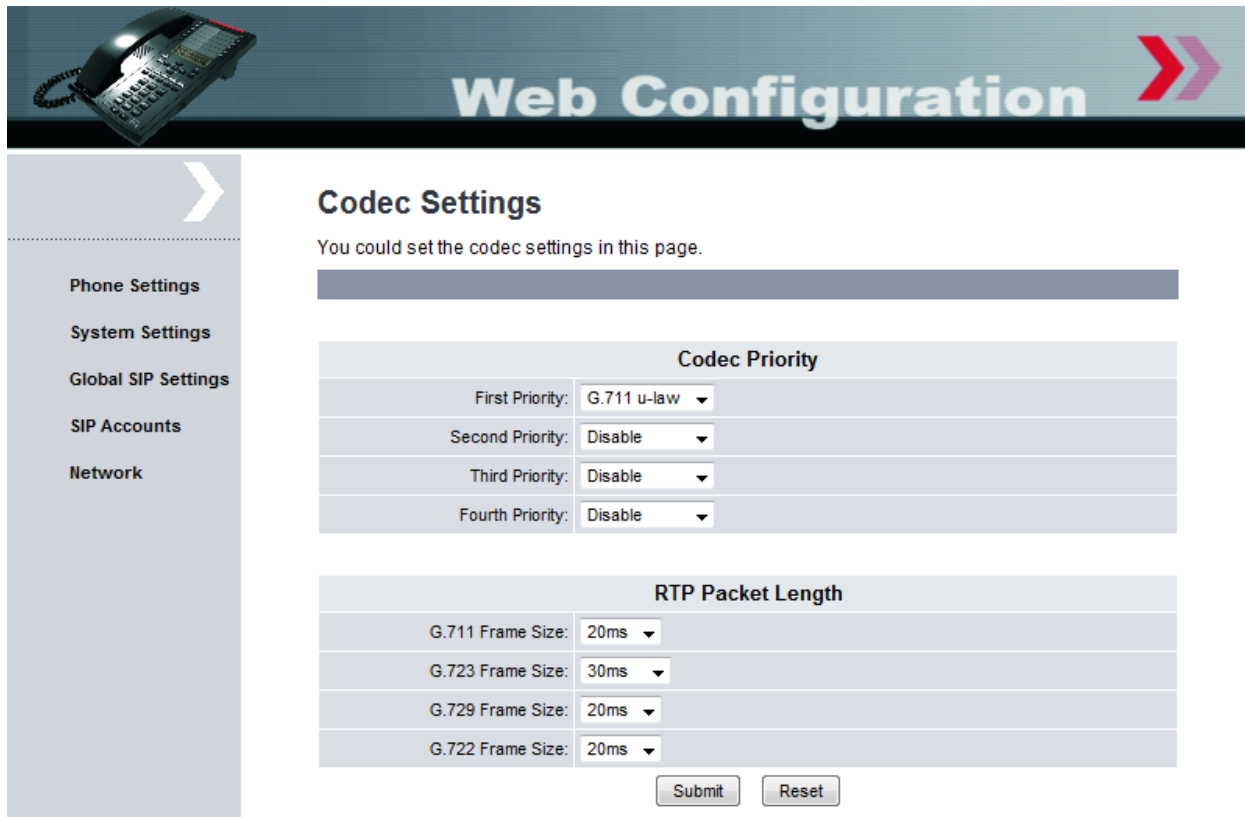
SIP Account 1	
Registration:	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
Registration ID:	50090
Display Name:	50090
Authentication Name:	50090
Password:	....
Registration Server:	dplab.com:5070
Proxy Server:	47.248.100.237:5070
Realm Address:	47.248.100.237:5070
BLF LIST:	
Voice Mail:	
Pickup Code:	
Expire Time:	300
DTMF Type:	RFC2833
Ping before register:	Disable
MWI:	Disable
Send KeepAlive:	<input type="radio"/> On <input checked="" type="radio"/> Off
Status:	

At the bottom of the form are two buttons: "Submit" and "cancel".

Figure 18 – G-Tek GE-202 SIP Account Settings

## 5.2. Codec settings

Figure 19 shows the codec settings for the SIP phone.



The screenshot shows the 'Web Configuration' interface for a G-Tek GE-202 SIP phone. On the left is a sidebar with navigation links: Phone Settings, System Settings, Global SIP Settings, SIP Accounts, and Network. The main content area is titled 'Codec Settings' and includes a sub-header 'You could set the codec settings in this page.' Below this are two configuration sections: 'Codec Priority' and 'RTP Packet Length'. The 'Codec Priority' section has four rows for First, Second, Third, and Fourth Priority, each with a dropdown menu. The 'RTP Packet Length' section has four rows for G.711, G.723, G.729, and G.722 Frame Sizes, each with a dropdown menu. At the bottom right are 'Submit' and 'Reset' buttons.

Codec Priority	
First Priority:	G.711 u-law
Second Priority:	Disable
Third Priority:	Disable
Fourth Priority:	Disable

RTP Packet Length	
G.711 Frame Size:	20ms
G.723 Frame Size:	30ms
G.729 Frame Size:	20ms
G.722 Frame Size:	20ms

Submit Reset

Figure 19 – G-Tek GE-202 Codec Settings

## 6. General Test Approach and Test Results

The focus of this interoperability compliance testing was primarily to verify the call establishment on the GE202 phones and the feature operations such as: busy, hold, DTMF, MWI, codec negotiation, bridge line appearance and dual call.

### 6.1. General test approach

The general test approach was to have one of the CS1000 clients/users to place a call to and from the G-Tek GE-202 and exercise the telephony features. The main objectives were to verify the GE-202 successfully perform the following:

- Register to CS1000- SIP Line system domain.

- Call establishment with Avaya CS1000 SIP and non SIP phones/clients
- Call establishment with emulated PSTN phones.
- Basic call operation: DTMF transmission, voicemail with MWI notification, busy, hold.
- Advance CS1000 Call Server features: speed dial, group call pickup, ring again busy/no answer, call park/retrieve (from Avaya phones), call forward (busy/all call/no answer), multiple line appearances and multiple appearances DN
- Call redirection and conference: Avaya phones as a transferor for blind/consultative transfers and as a moderator for the 3 way conference call.
- Specific hospitality feature requirement; bridge line appearance (BLA) and dual call arrangement
- Redirect call between users/clients/endpoints: bridge line appearance and dual call
- Codec negotiations.

## 6.2. Test Results

The objectives outlined in section 6.1 were verified and met.

The following observations were made during the compliance testing:

- Avaya has not performed audio performance testing or reviewed the G-Tek GE202 compliance to required industry standards
- Enable Network Call Redirection (NCRD) in CS1000 Call Server SIP Line Route will cause an issue with Call Waiting. CR Q02073088 has been raised against CS1000 SIP Line system. This will be fixed in a future release of CS1000.
- GE202 supports only G711 for this version of firmware.

## 7. Verification Steps

This section includes some steps that can be followed to verify the configuration.

- Verify that the VVX1500 phone registers successfully with the CS1000 SIP Line Gateway server and Call Server by using CS1000 Linux command line and CS1000 Call Server overlay LD 32

- Login to the sipline server using the nortel account.
- Issue command “slgSetShowByUID [userID]” where userID is SIP Line user’s ID being checked

```
[nortel@sipl ~]$ slgSetShowByUID 55524
=== VTRK ===
UserID          TN          Clients  Calls  SetHandle
-----
55524          096-00-02-24          1      0  0xb7c25e10
StatusFlags = Registered Controlled KeyMapDwld SSD
FeatureMask =
CallProcStatus = 0

Current Client = 0, Total Clients = 1
```

```

== Client 0 ==
IP:Port:Trans = 47.248.100.56:5060:udp
Type          = SIP3
UserAgent     = PolycomV VX-VVX_1500-UA/3.2.2.0481
x-nt-guid     = 9ad7a4871842718a35aeadf070608745
RegDescrip    =
RegStatus     = 1
PbxReason     = OK
SipCode       = 200
Expire        = 300
Contact       = sip:55524@47.248.100.56:5060
Nonce         = cad64489bbd9bca62aa9a1f833052da4
NonceCount    = 3
hTimer        = 0x9c659d0
TimeRemain    = 183
Stale         = 0
Outbound      = 0
ClientGUID    = 0

```

Key	Func	Lamp	Label
0	3	0	55524
1	126	0	2655524
2	3	0	55097
3	9	0	
4	29	0	
17	16	0	
18	18	0	
19	27	0	
20	19	0	
21	52	0	
22	25	0	
24	11	0	
25	30	0	
26	31	0	

- Login to the call server using the admin account.
- Load overlay 32 and then issue command “stat [TN]” where TN is the SIP Line user’s TN being checked

```

>ld 32
NPR000
.stat 96 0 2 24
IDLE REGISTERED 00

```

- Place a call from and to the GE202 phone and verify that the call is established with 2 way speech path.
- During the call, use pcap tool (ethereal/wireshark) at the SIPLine Gateway and clients to make sure that all SIP request/response messages are correct.

## 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 G-Tek

and Avaya design teams. Some of these issues are considered as exceptions. The SIP phone GE202 version 1820X.27.1.01 is considered compliant with CS1000 SIP Line System Release 6.0.

## 9. Additional References

Product documentation for Avaya products may be found at:

<http://support.nortel.com/go/main.jsp>

[1] *Communication Server 1000 SIP Line Fundamental, Release 6.0, Revision 01.08, February 2010, Document Number NN43001-508*

[2] *Communication Server 1000E Maintenance, Release 6.0, Revision 03.16, January 2010, Document Number NN43041-700*

[3] *Communication Server 1000 ISDN Primary Rate Interface Installation and Commissioning, Revision 01.03, August 2007, Document Number NN43001-301*

[4] *Troubleshooting Guide for Distributors, Release 6.0, Revision 02.02, December 2009, Document Number NN43001-730*

[5] *Communication Server 1000E Installation and Commissioning, Release 6.0, Revision 03.06, February 2010, Document Number NN43041-310*

[6] *Communication Server 1000E Software Upgrades, Revision 03.12, February 2010, Document Number NN43041-458*

[7] *Communication Server 1000E Linux Platform Base and Applications Installation and Commissioning, Revision 03.10, February 09, 2009, Document Number NN43001-315*

[8] *Communication Server 1000 Unified Communications Management Common Services Fundamentals, Revision: 03.04, September 28, 2009, Document Number NN43001-116*

Product information for G-Tek products can be found at

[http://www.G-Tek.com.tw/en/offering\\_products.php?mw=9](http://www.G-Tek.com.tw/en/offering_products.php?mw=9)

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