Avaya Solution & Interoperability Test Lab

Application Notes for Avaya B179 SIP Conference Phone with Avaya Communication Server 1000 Release 6.0 – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Server 1000 Release 6.0 and the Avaya B179 SIP Conference Phone. The B179 is a SIP VoIP conference Telephone that registers as a 3rd Party SIP Line client with Communication Server 1000 Release 6.0. The solution supports calling among the B179 and other Communication Server 1000-supported non-SIP and SIP Line clients.

Testing was conducted by the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe a solution comprised of Avaya Communication Server 1000 Release 6.0 and the Avaya B179 SIP Conference Phone. The B179 is a SIP VoIP conference telephone that registers as a 3rd Party SIP Line client with Communication Server 1000 Release 6.0. This solution supports calling among the B179 and other Communication Server 1000-supported non-SIP and SIP Line clients.

Figure 1 illustrates the network configuration of equipment that was used for testing. All telephones, including the B179, are registered to Avaya Communication Server 1000 release 6.0. The telephones were configured in the 55xxx extension range.

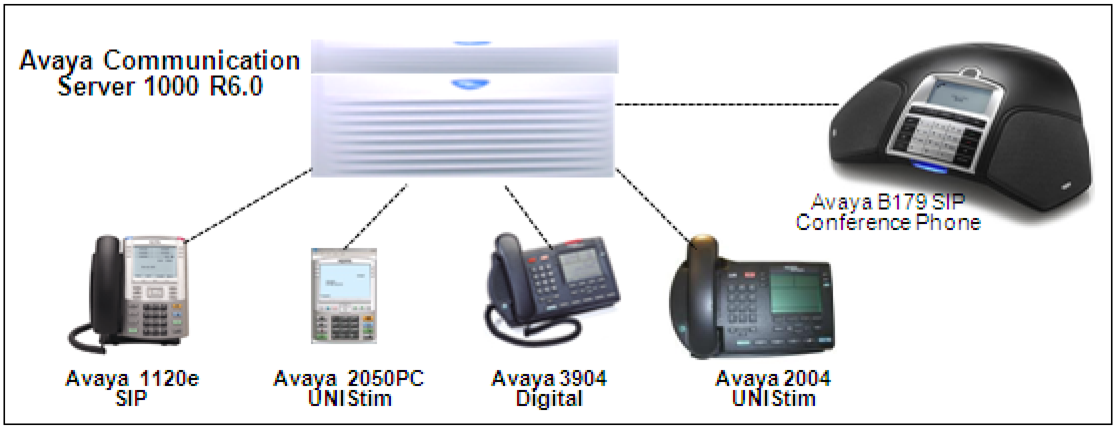


Figure 1: Network Configuration

2. Equipment and Software Validated

Provider	Hardware Component	Software Version
Avaya	Avaya Communication Server 1000E CPPM	VERSION 4021 RELEASE 6 ISSUE 00 R +
Avaya	Avaya 1140e IP Desk phone	SIP: 04.01.13.00
Avaya	Avaya 2004 IP Desk phone	UNIStim: 0622B76
Avaya	Avaya IP Softphone 2050PC	UNIStim: 4.01.041
Avaya	Avaya M3904 Digital Phone	N/A
Avaya	Avaya B179 SIP Conference Phone	2.2 and 2.2.1

Table 1: Hardware Components and Software Versions

Update Type	Update Components
Deplist	DepList 1: core Issue: 03 (created: 2011-04-26 15:23:48 (est))
Service Packs	In system patches: 6 NAME RPM p28774_1 nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386 p28797_1 nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386 p29703_1 nortel-cs1000-shared-ssSubagent-6.00.18-00.i386 p28961_1 nortel-cs1000-pi-control-1.00.00.00-00.noarch p30043_1 nortel-cs1000-OS-1.00.00.00-00.noarch p30274_1 nortel-cs1000-Jboss-Quantum-6.00.18.00-00.i386 In System service updates: 11 NAME nortel-cs1000-shared-pbx-6.00.18.065-01.i386.002 nortel-cs1000-patchWeb-6.00.18.65-01.i386.001 nortel-cs1000-vtrk-6.00.18.65-TMP410.i386.000 nortel-cs1000-dmWeb-6.00.18.62-00.i386.001 nortel-cs1000-ISECSH-6.00.18.62-00.i386.000 ntp-4.2.4p8-1.el5.pp.i386.000 tzdata-2009u-1.el5.noarch.000 nortel-cs1000-csv-6.00.18.65-04.i386.000 nortel-cs1000-linuxbase-6.00.18.65-06.i386.000 nortel-cs1000-auth-6.00.18.65-01.i386.000 nortel-cs1000-shared-general-6.00.18.62-01.i386.000

3. Configure Avaya Communication Server 1000

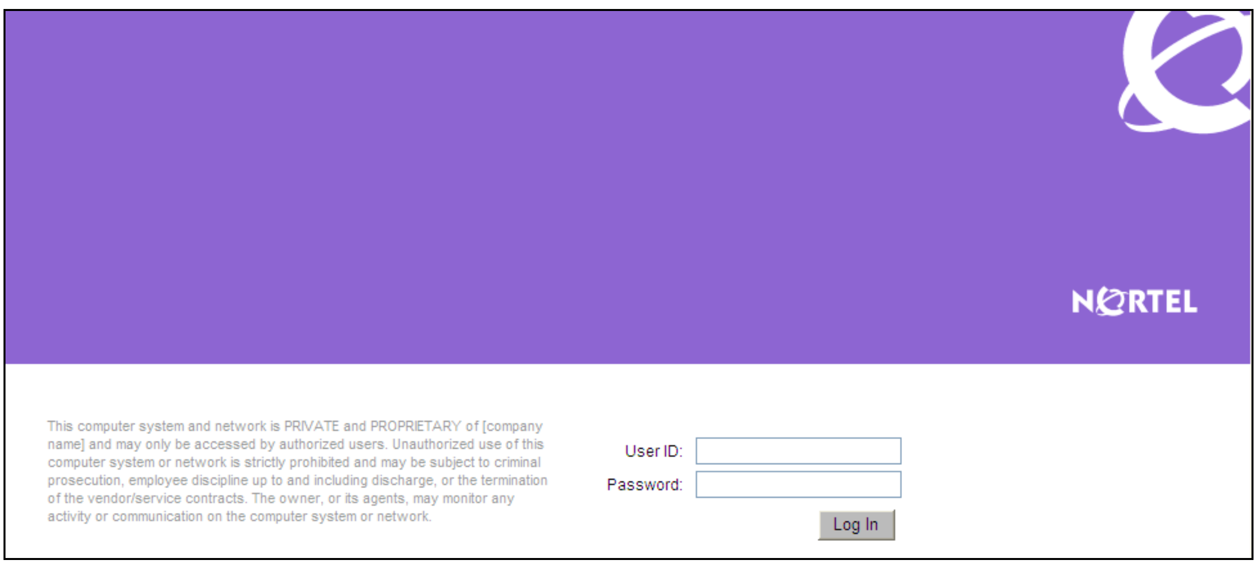
This section describes the steps to configure the following, using CS 1000 Element Manager:

- SIP Line service
- SIP Line D-Channel
- Application Module Link (AML)
- Value Added Server (VAS)
- Zone for SIP phones
- SIP Line Route Data Block (RDB)
- SIP Line Virtual Trunk
- Media Gateway Controller
- SIP Line telephone corresponding to the B179 SIP Conference Phone

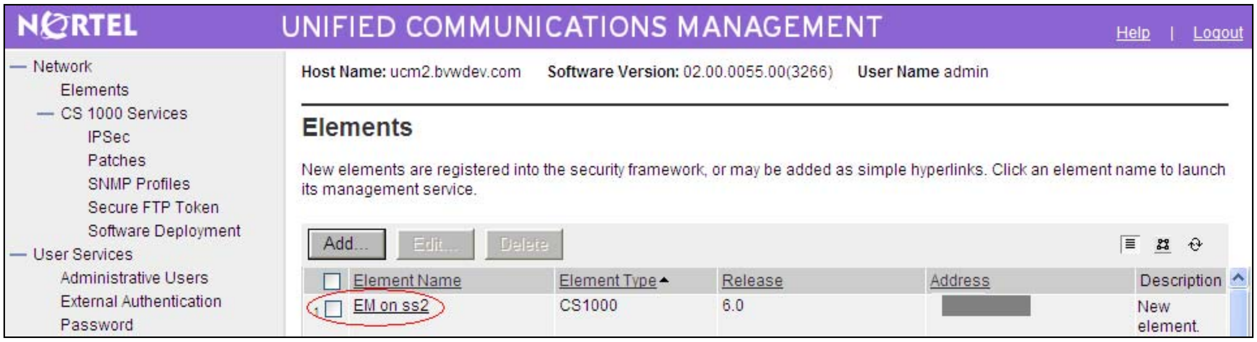
It is assumed that basic installation and configuration of the CS 1000 call server, signaling server, and node have been completed. Additional configuration details are provided in **[1, 2]**.

3.1. Log in to Element Manager (EM)

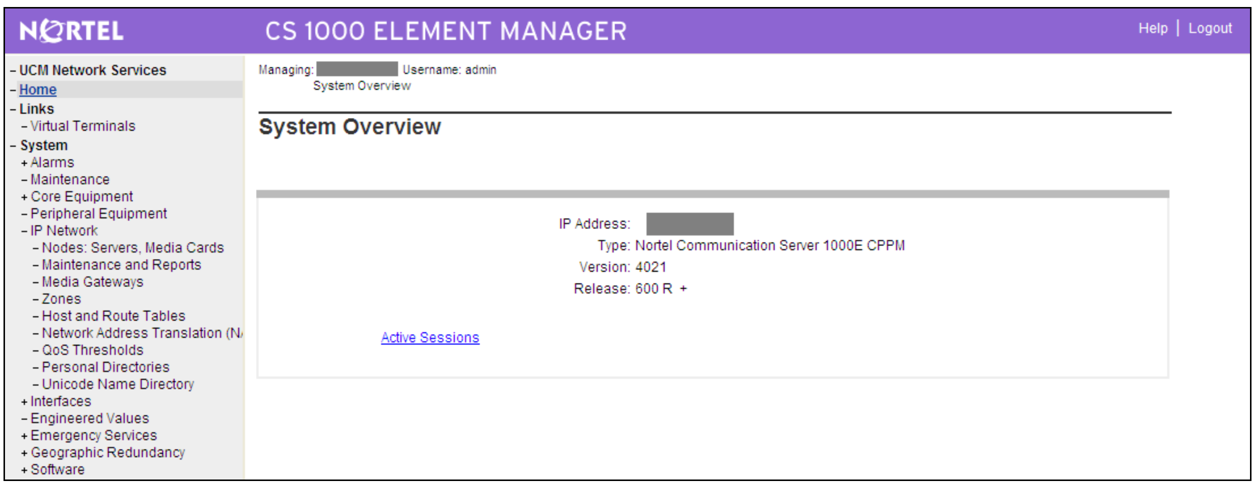
Access the Unified Communications Management (UCM) web based interface by using the URL “http://<ip-address>” in an Internet browser window, where “<ip-address>” is the IP address of the UCM server.
Log in with the appropriate user name and password.



The following **Unified Communications Management** screen will be displayed. Click on the **Element Name** corresponding to the **Element Manager (EM)**.

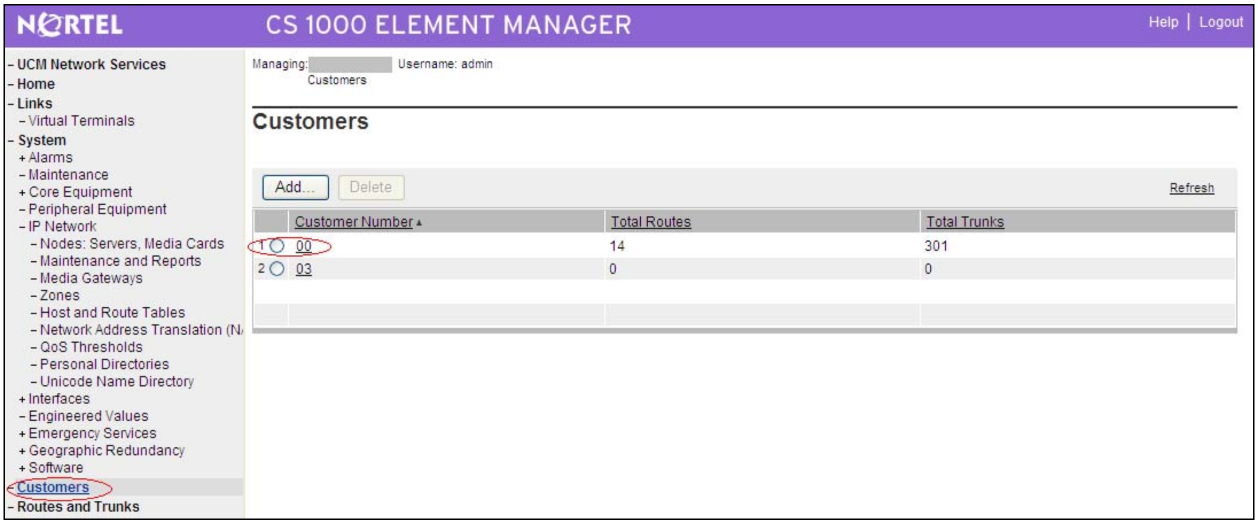


The CS 1000 Element Manager page appears as shown below.

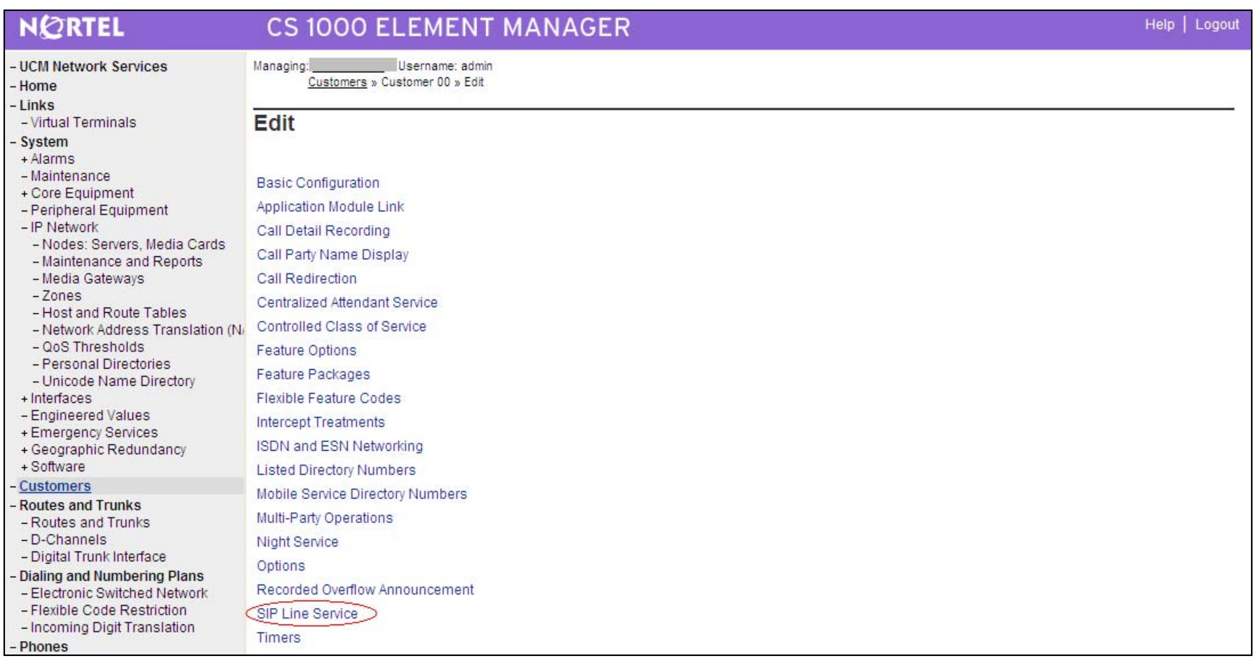


3.2. Enable SIP Line Service

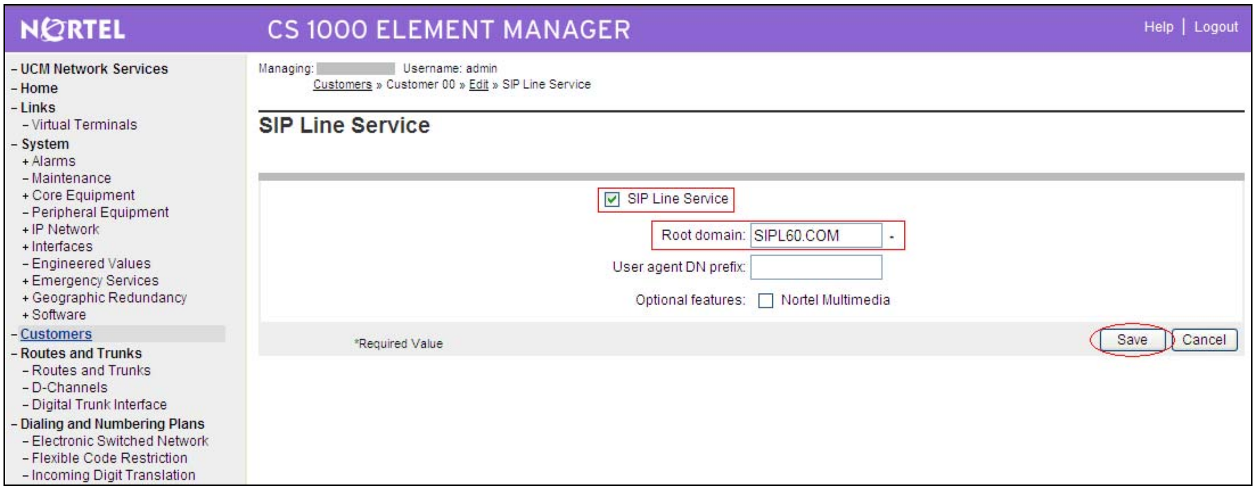
Select **Customers** in the left pane. The **Customers** screen is displayed. Click the link associated with the appropriate customer, in this case **00**. The system can support more than one customer with different network settings and options. In the sample configuration, only one customer was configured on the system.



The **Customer Details** screen is displayed next. Select **SIP Line Service** to edit its parameters.

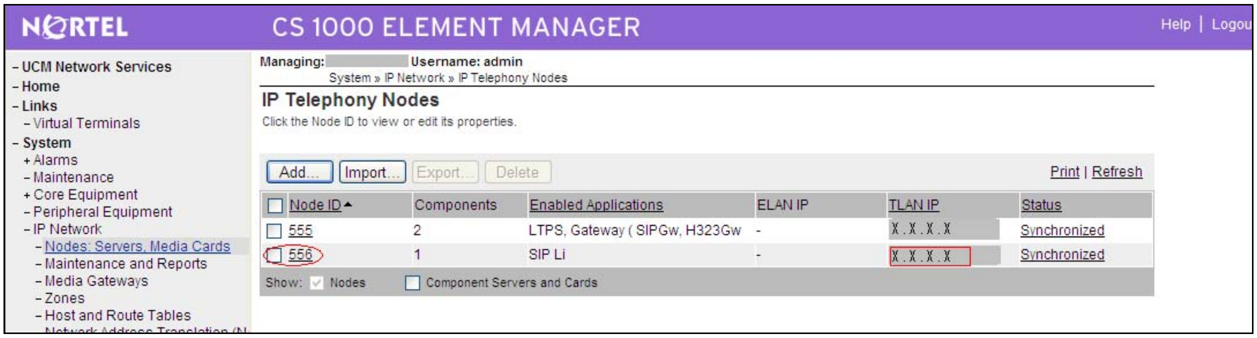


Check the **SIP Line Service** checkbox, enter an appropriate **Root domain**, and click **Save**.

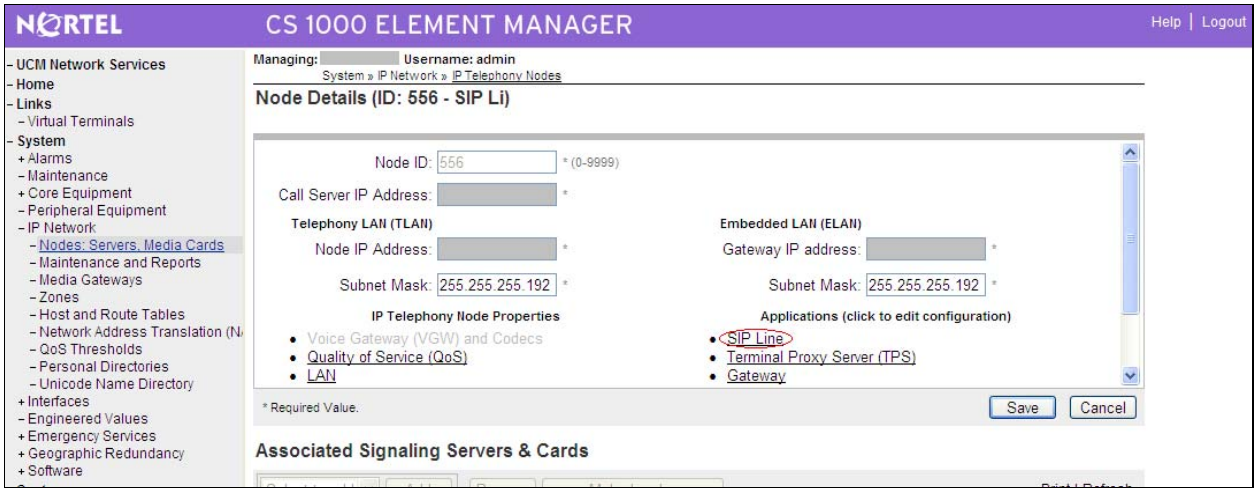


3.3. Enable SIP Line Service on Telephony Node

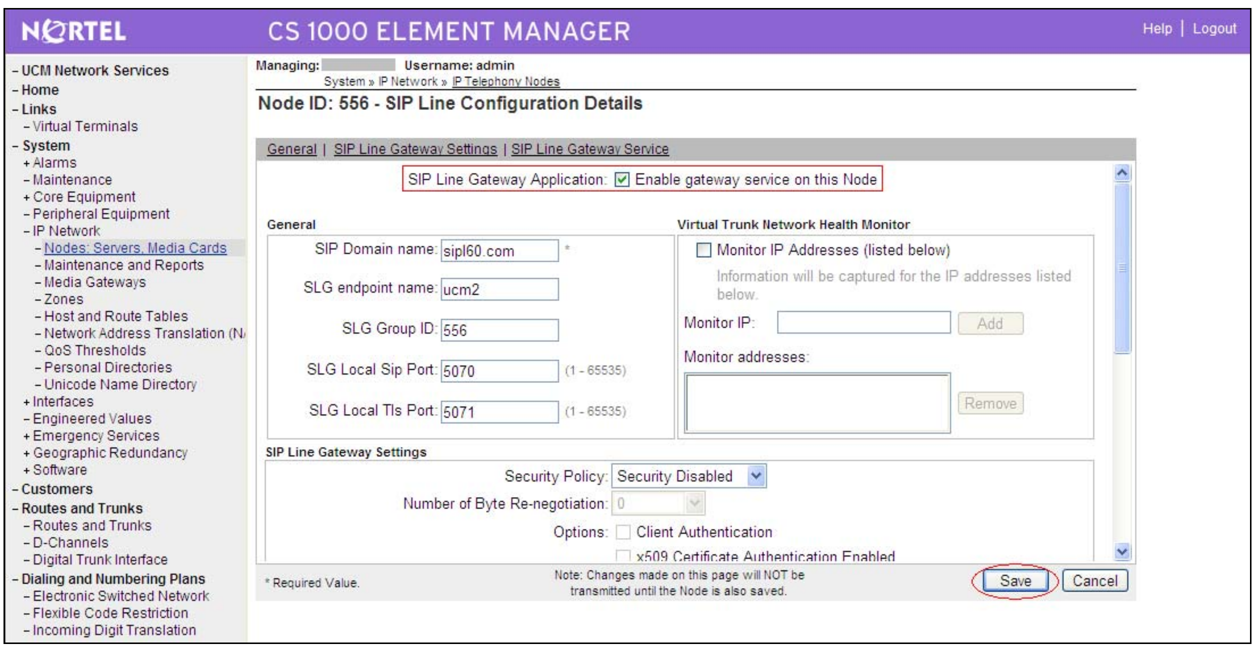
On the Element Manager page, navigate to **System → IP Network → Nodes: Servers, Media Cards**. Note the IP address of the SIP Line Node, as it will be used in configuring the B179 later. It would be displayed where X.X.X.X is indicated below. Select the **Node ID** on which SIP Line service is to be enabled.



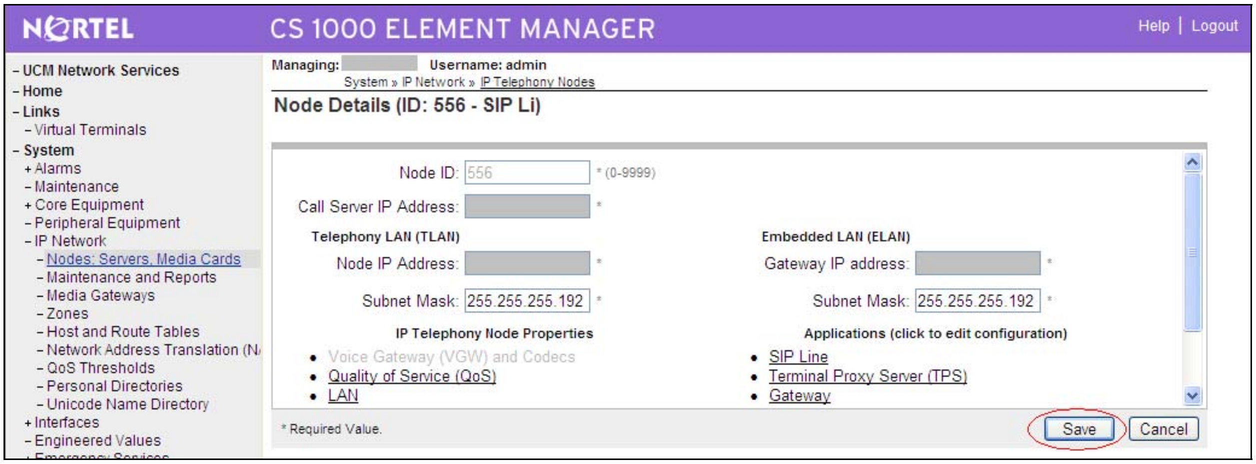
Scroll down the top section to display the **Applications** section on the right, and click on **SIP Line**.



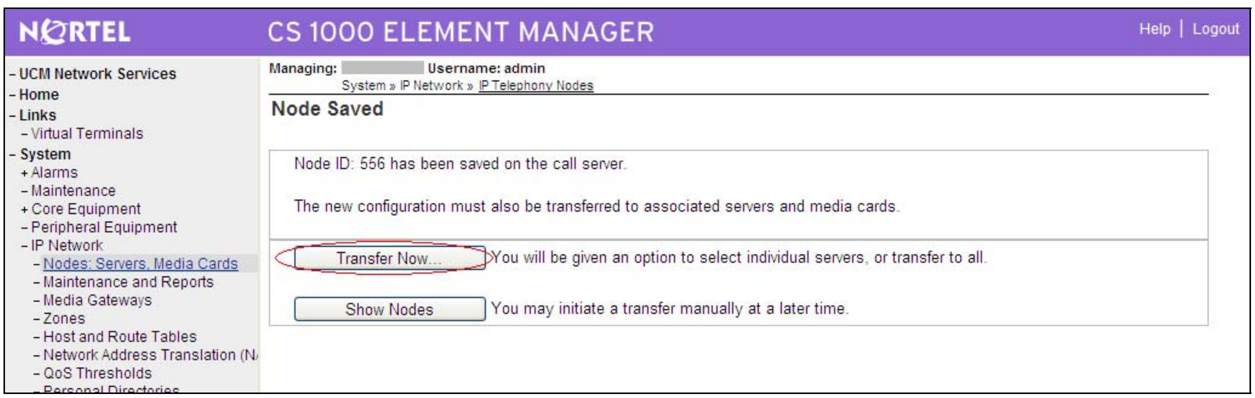
The SIP Line Configuration Details page is displayed. Check **Enable gateway service on this node** next to **SIP Line Gateway Application:** Then click **Save**.



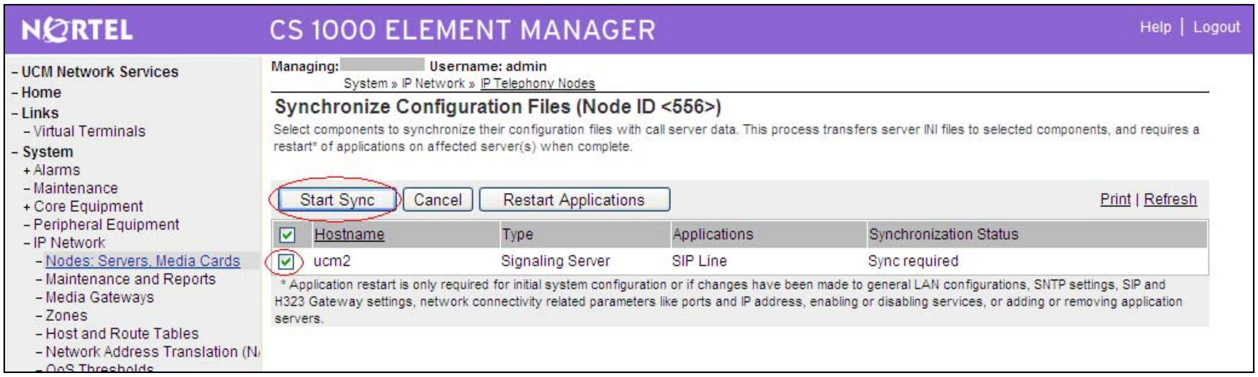
The **Node Details** screen then returns. Click the **Save button** on this screen.



Select **Transfer Now** on the **Node Saved** page as shown below.

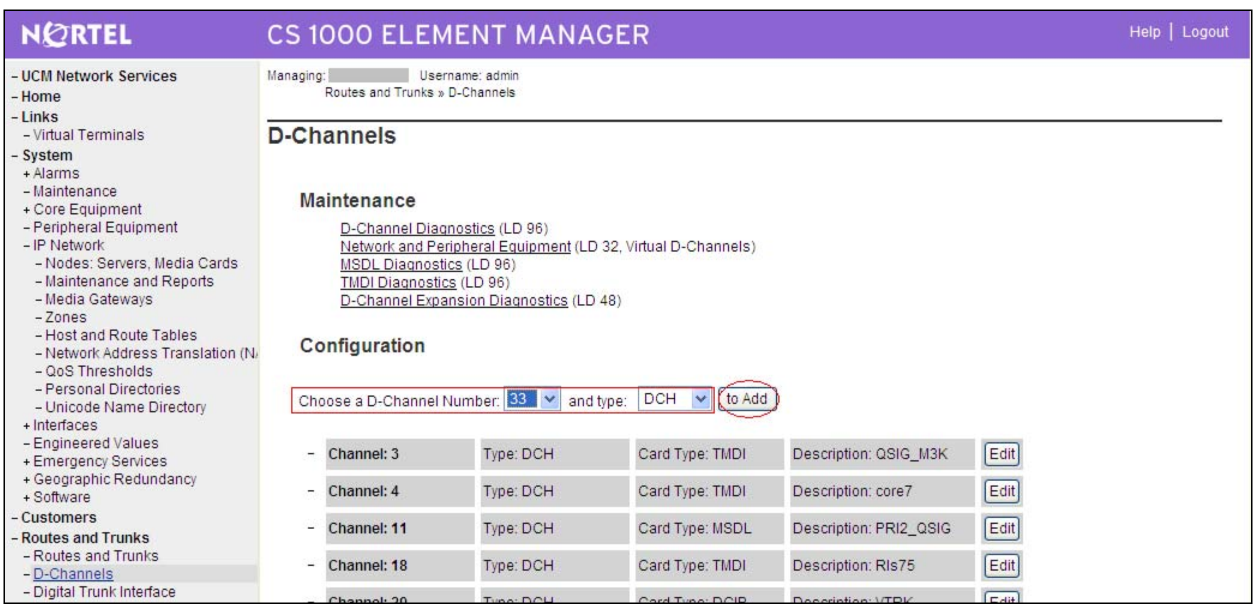


Once the transfer completes, the **Synchronize Configuration Files (Node ID <id>)** page is displayed as shown below. Check the appropriate SIP Line Server and click **Start Sync.** The screen will automatically refresh until the synchronization is finished. The **Synchronization Status** field will update from **Sync required** (as shown) to **Synchronized** (not shown). After synchronization completes, click **Restart Applications** to use the new SIP Gateway settings.

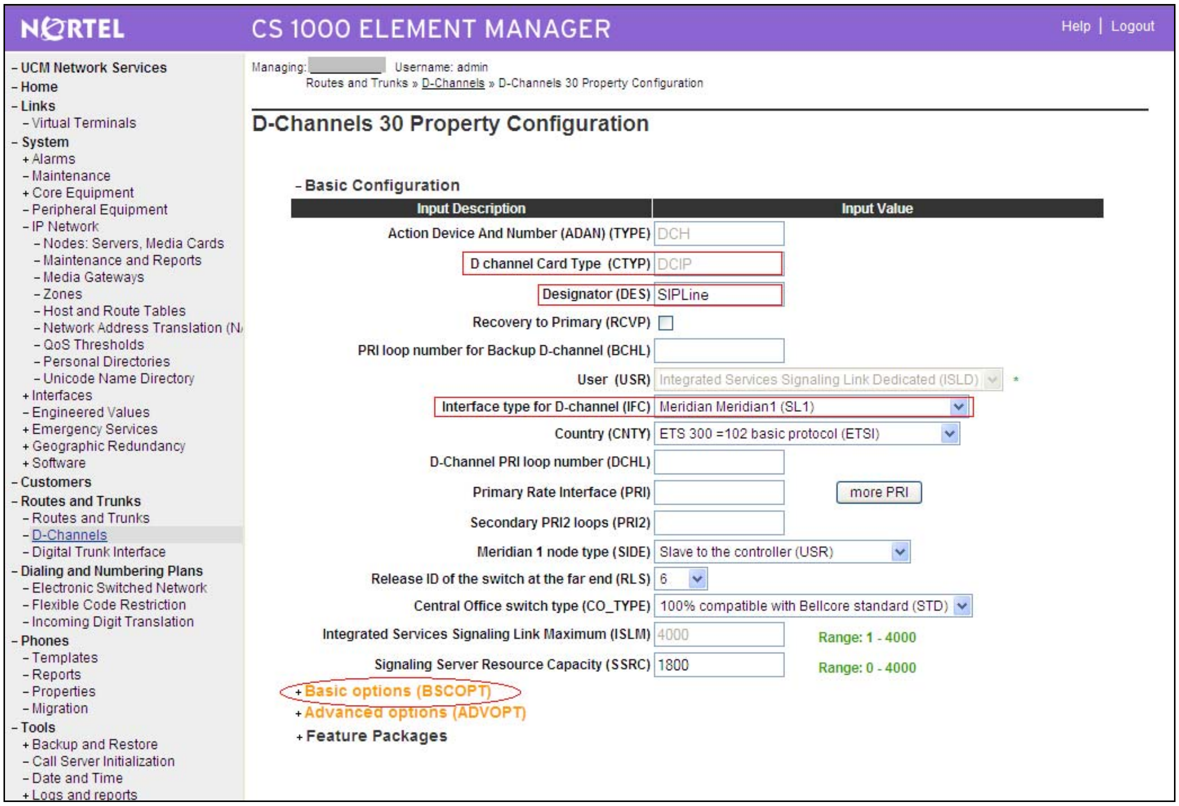


3.4. Configure SIP Line D-Channel

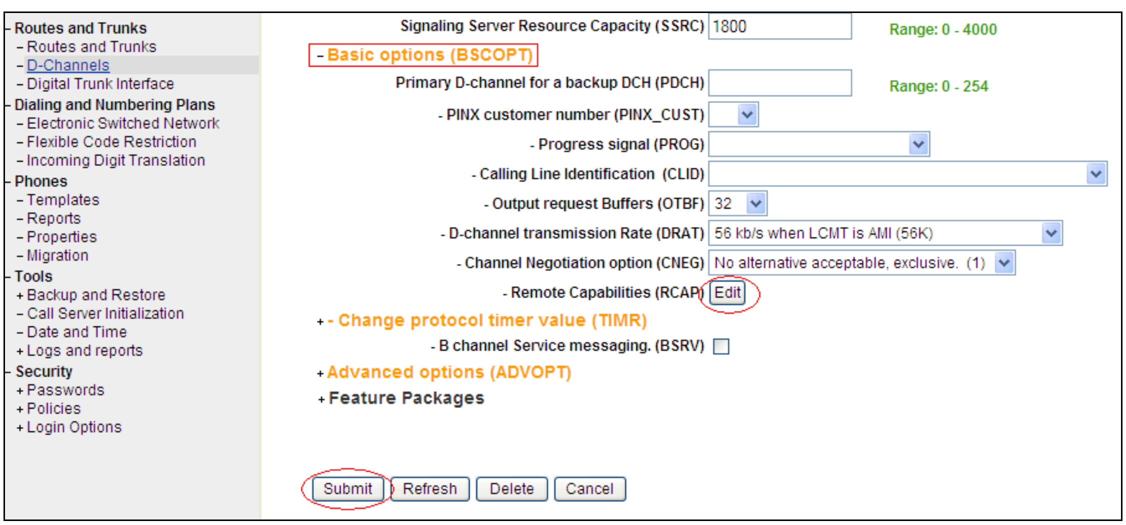
On the left column menu of the main Element Manager page, navigate to **Routes and Trunks → D-Channels**. Under the **Configuration** section, select a D-Channel number from the **Choose a D-Channel Number** list (channel 33 in the sample configuration), and select **DCH** for the **type**. Click **to Add**.



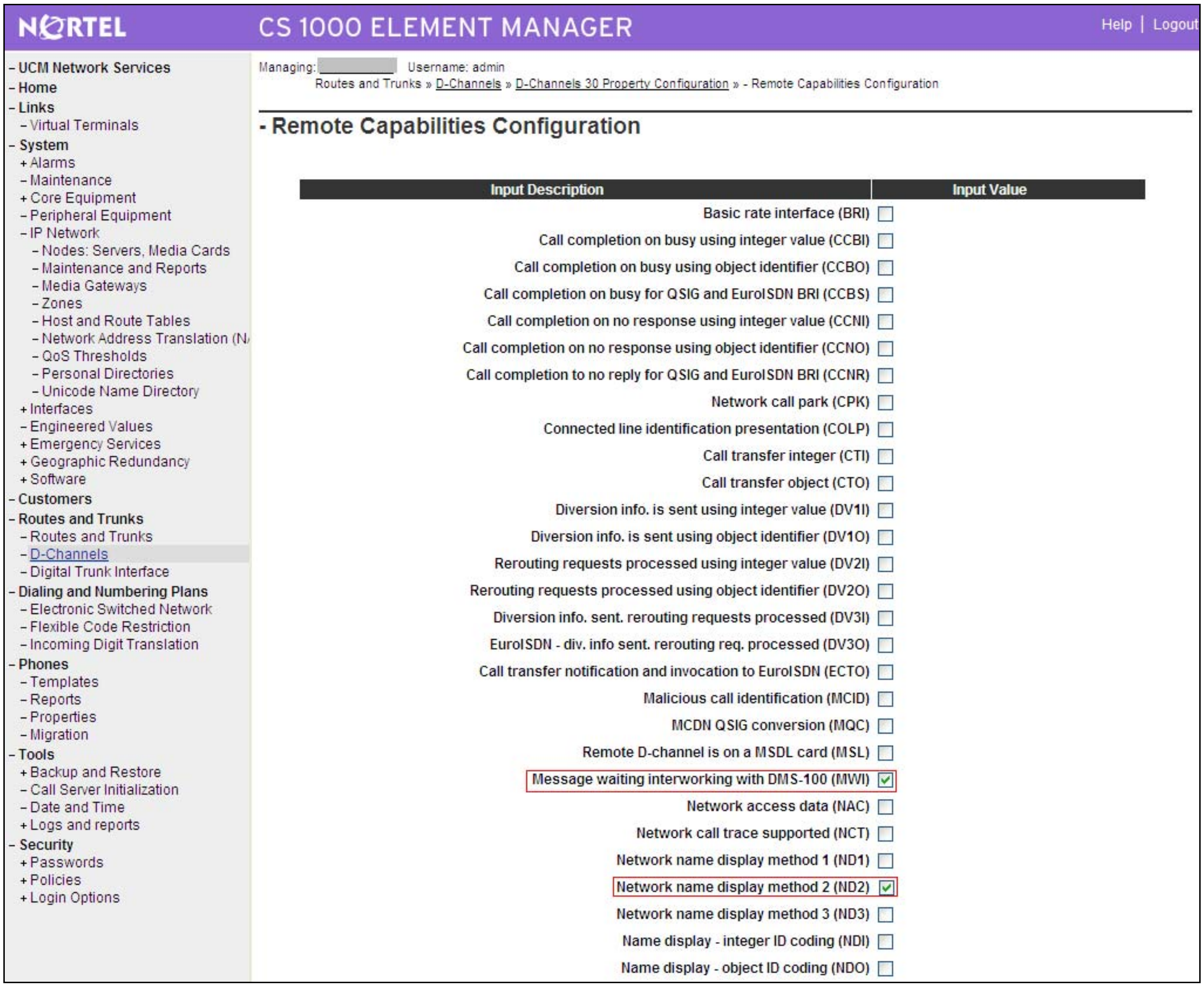
The **D-Channels Property Configuration** screens below show the parameter values after configuring the D-channel. **DCIP** is selected for **D channel Card Type**, **Meridian Meridian1 (SL1)** is selected for **Interface type for D-channel,** and an appropriate **Designator** is entered. The remaining parameters have their default values.



Click the **Basic options (BSCOPT)** link to expand that section. Click **Edit** to configure **Remote Capabilities.**



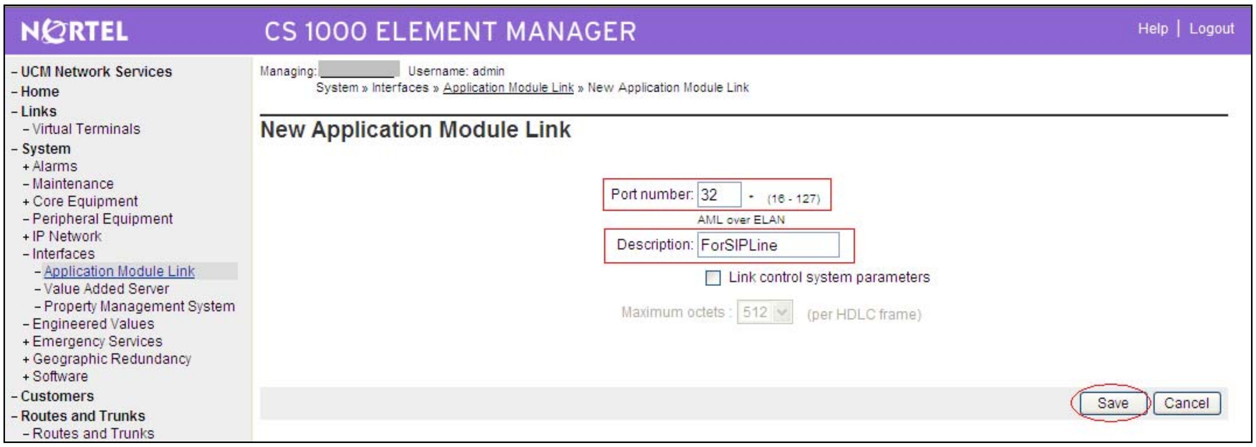
The **Remote Capabilities Configuration** page is displayed as shown below. Select the **Message waiting interworking with DMS-100 (MWI***)* check box,[1] and the **Network name display method 2 (ND2)** check box. At the bottom of the **Remote Capabilities Configuration** page, click **Return - Remote Capabilities** (not shown), and the **D-Channel Property Configuration** page reappears as shown in the previous screen. Click on **Submit** .



[1] Note that although the Avaya B179 SIP Conference Telephone does not support Message Waiting Indicator, this D channel can also be used for other SIP Line IP telephones that do support it, so it is enabled here for that purpose.

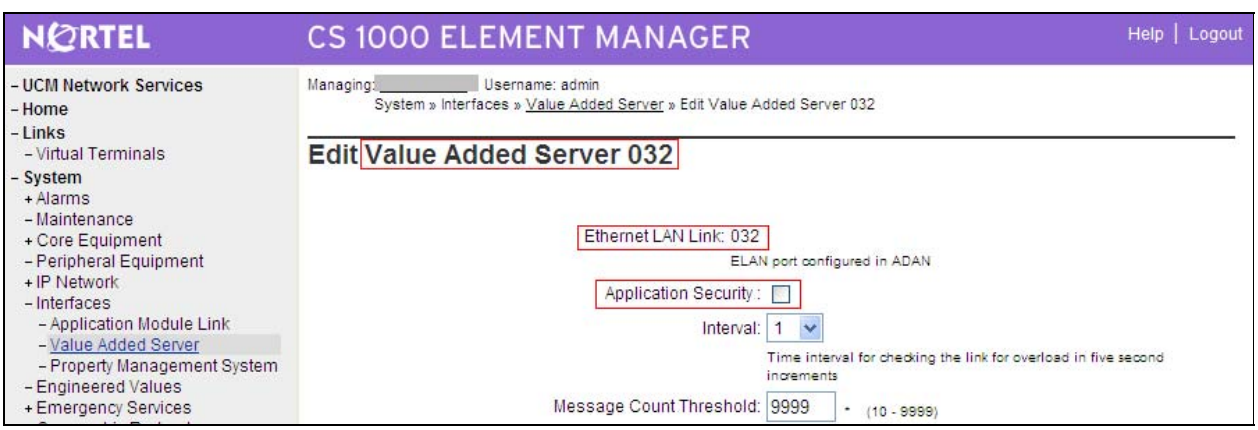
3.5. Configure Application Module Link (AML)

On the left column menu of the main Element Manager page, navigate to **System → Interfaces → Application Module Link**, and click **Add** (not shown). The **New Application Module Link** page is displayed. Enter the AML port number in the **Port number** text box. The SIP Line Service can use ports 32 through 127. In the sample configuration, the SIP Line Service is configured to use port 32. Enter an appropriate **Description**. Click **Save** to save the configuration.



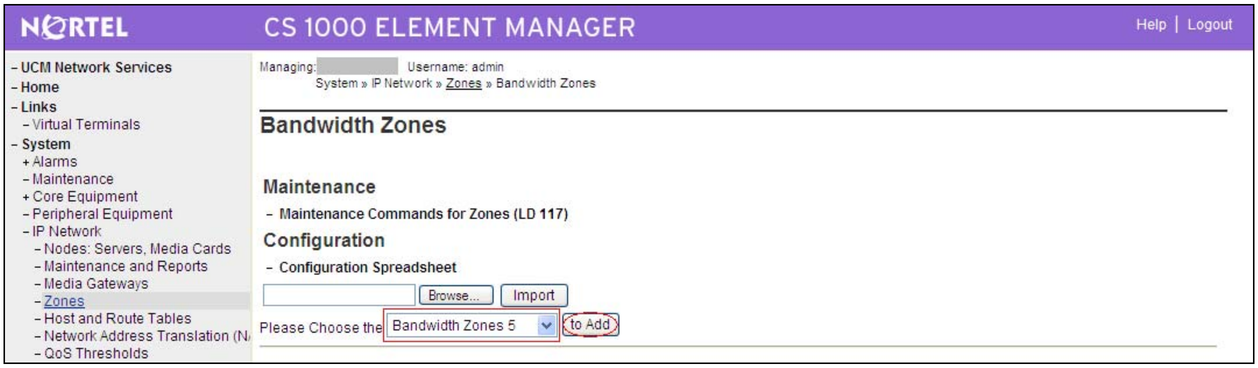
3.6. Configure Value Added Server (VAS)

On the left column menu of the main Element Manager page, navigate to **System → Interfaces → Value Added Server**. Click **Add** and then click **Ethernet LAN Link** on the **Add Value Added Server** page that is displayed next (not shown). On the **Ethernet Link** page that is displayed next, enter a **Value added server ID** (32 in the sample configuration), and select the AML number created in the previous section for **Ethernet LAN Link**. Ensure that the **Application Security** check box is unchecked. Click **Save** (not shown). The screen below shows the result of adding the value added server.

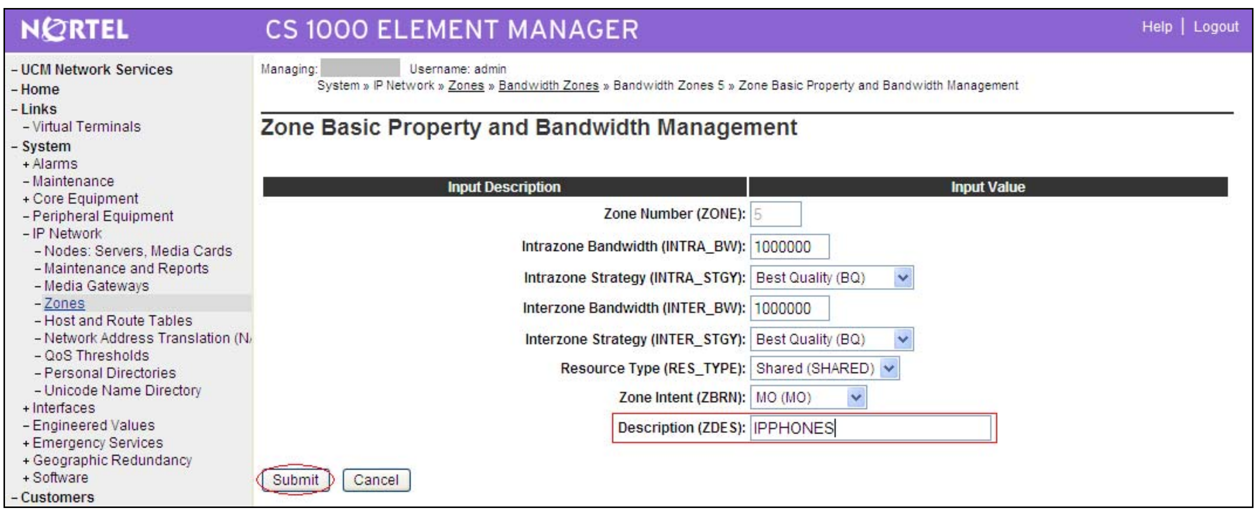


3.7. Configure Zone for SIP Phones

On the left column menu of the main Element Manager page, navigate to **System → IP Network → Zones**. On the **Zones** page, select **Bandwidth Zones** (not shown). On the **Bandwidth Zones** page, choose a new Bandwidth Zone from the drop-down box and click on **to Add**.

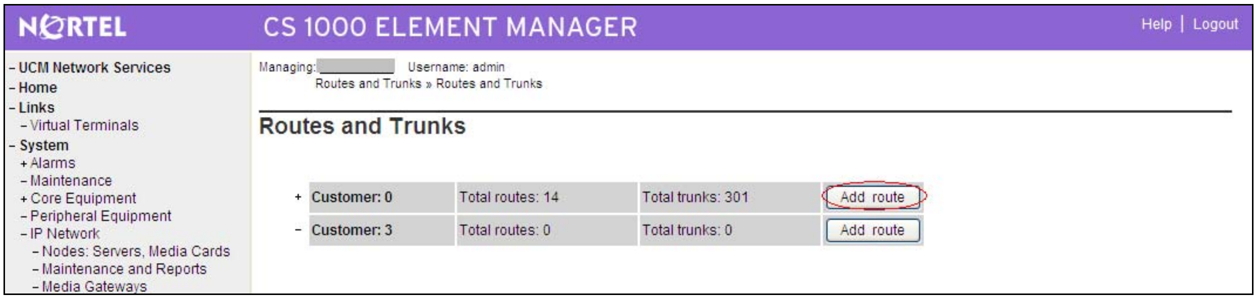


On the **Zone Basic Property and Bandwidth Management** page, enter an appropriate **Description**. Defaults can be used for the remaining fields. Click **Submit**.



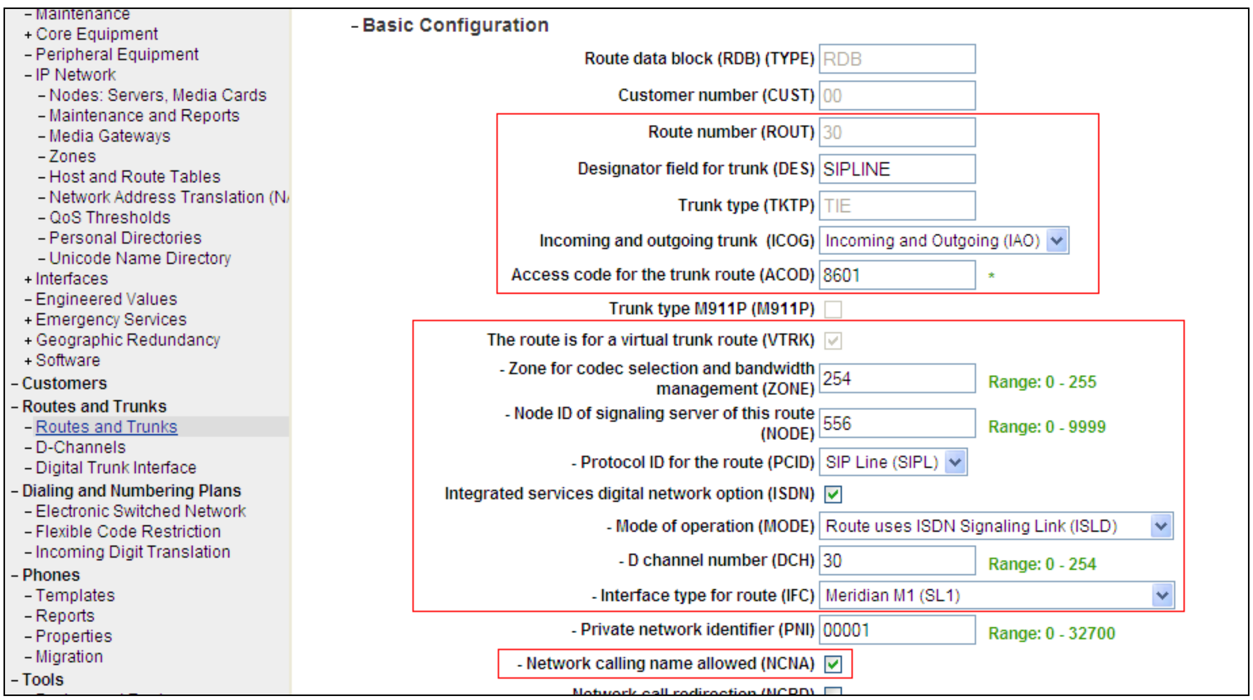
3.8. Configure SIP Line Route Data Block (RDB)

On the left column menu of the main Element Manager page, navigate to **Routes and Trunks → Routes and Trunks**. Click **Add route** for the appropriate customer number.



The following screen shows the parameter settings after the route has been added. Set the following parameters and leave default values for the remaining parameters. The **Basic Route Options**, **Network Options**, **General Options**, and **Advanced Configurations** sections (not shown) can be left at the defaults. Click **Submit** (not shown) to save the configuration changes.

Route number (ROUT)	Select the route number
Designator field for trunk (DES)	Enter an appropriate name
Trunk type (TKTP)	Select **TIE trunk data block (TIE)**
Incoming and outgoing trunk (ICOG)	Select **Incoming and Outgoing (IAO)**
Access code for the trunk route (ACOD)	Enter the access code
The route is for a virtual trunk route (VTRK)	Check the box
Zone for codec selection and bandwidth management (ZONE)	Enter a zone[2]
Node ID of signaling server of this route (NODE)	Enter the node ID of the SIP Line Gateway
Protocol ID for the route (PCID)	Select **SIP Line (SIPL)**
Integrated services digital network option (ISDN)	Check the box
Mode of operation (MODE)	Select **Route uses ISDN Signaling Link (ISLD)**
D channel number (DCH)	Enter the D-channel number
Interface type for route (IFC)	Select **Meridian M1 (SL1)**
Network calling name allowed (NCNA)	Check the box

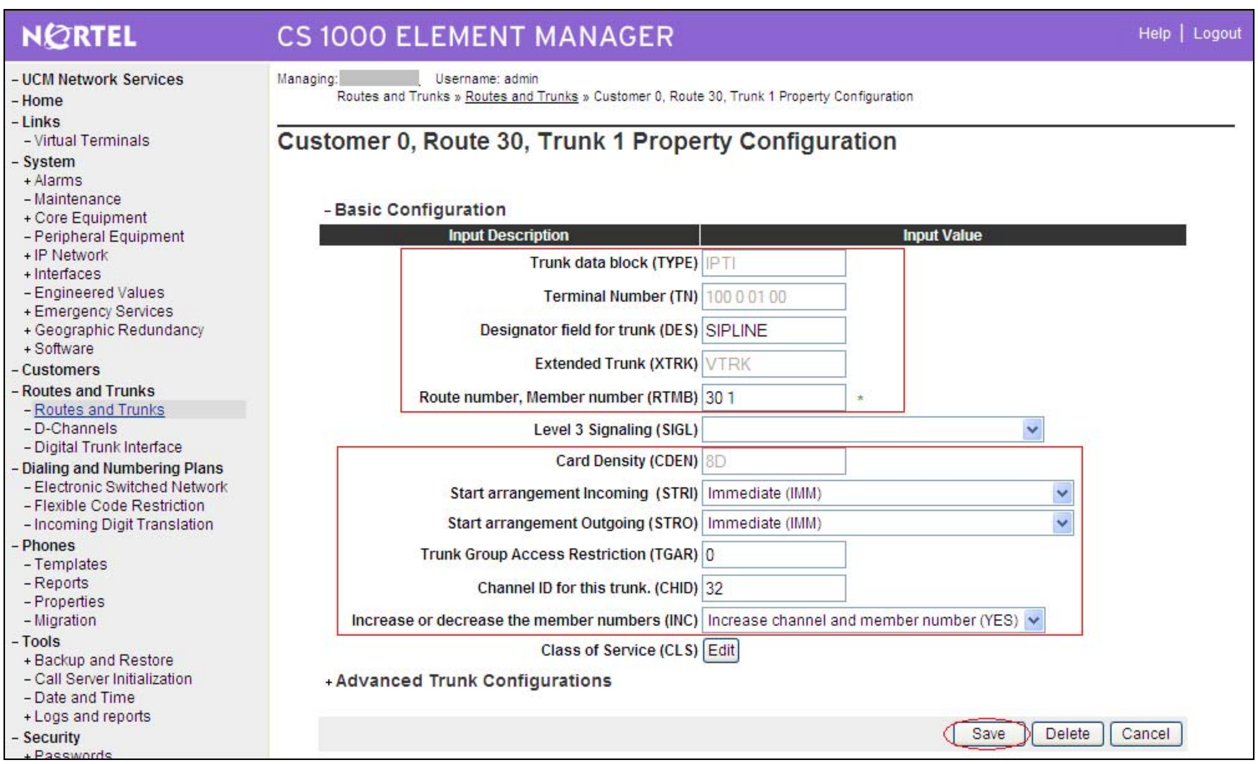


[2] Note that this must be a zone of type VTRK and must be different than the zone created for the SIP phones in **Section 3.7**. In the sample configuration, the VTRK zone was 254.

3.9. Configure SIP Line Virtual Trunk

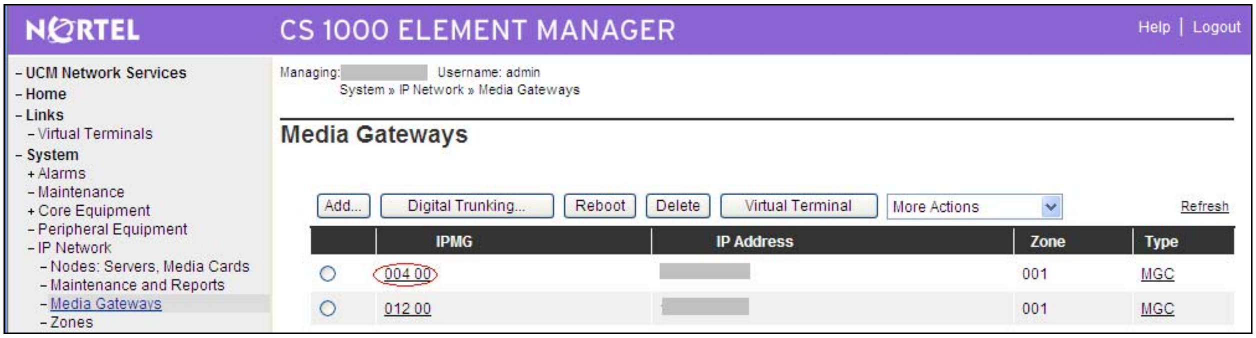
When the **Routes and Trunks** screen is displayed after adding the route in **Section 3.8**, click **Add trunk** corresponding to the newly added route to add new trunk members. The following screen shows the parameter settings for one of the trunks after they have been added. Set the following parameters and leave default values for the remaining parameters. Click **Save** to save the configuration changes.

Multiple trunk input number	Enter the number of trunks (only shown when adding trunks)
Trunk data block	Select **IP Trunk (IPTI)**
Terminal Number	An available terminal number.
Designator field for trunk	A descriptive text.
Extended Trunk	Select **Virtual trunk (VTRK)**
Route number, Member number	Current route number and starting member. (only shown when adding trunks)
Card Density	Select **Octal Density (8D)**
Start arrangement Incoming	Select **Immediate (IMM)**
Start arrangement Outgoing	Select **Immediate (IMM)**
Trunk Group Access Restriction	Desired trunk group access restriction level.
Channel ID for this trunk	An available starting channel ID.

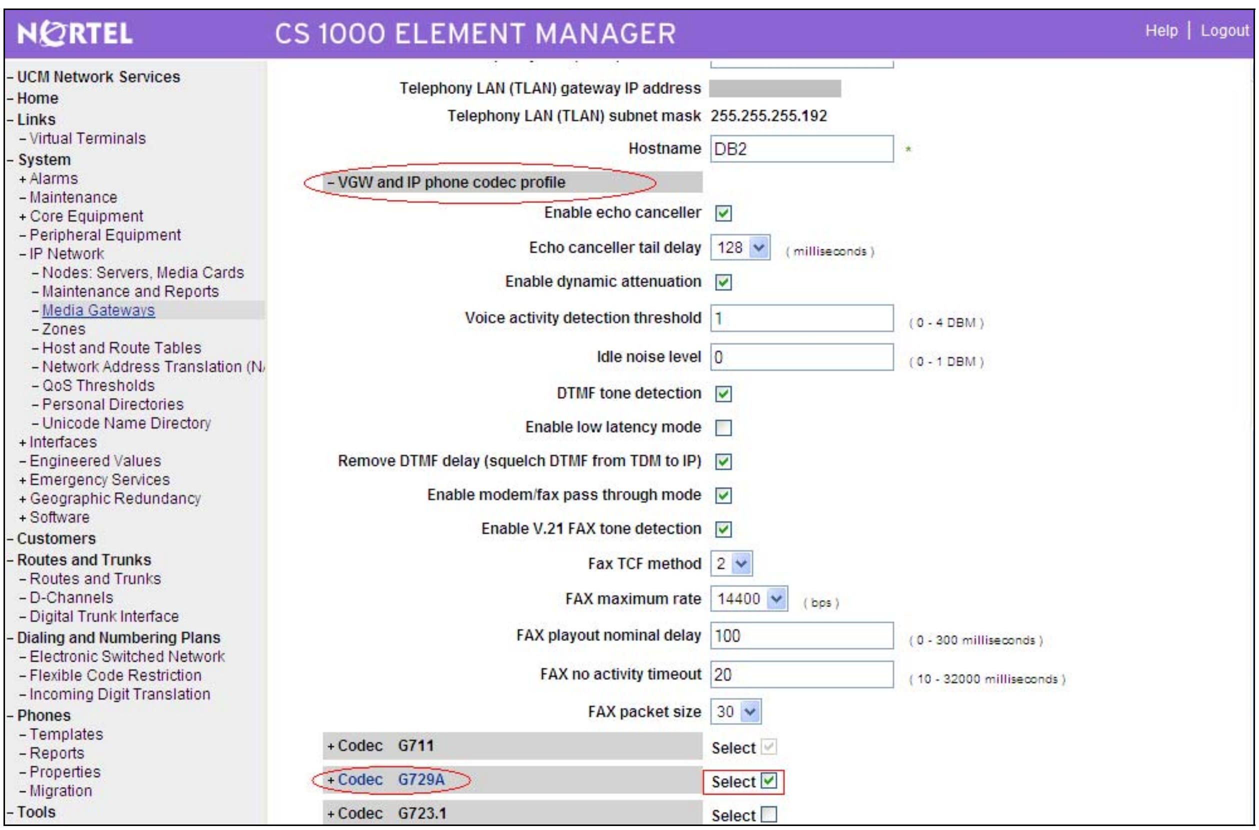


3.10. Configure Media Gateway Controller

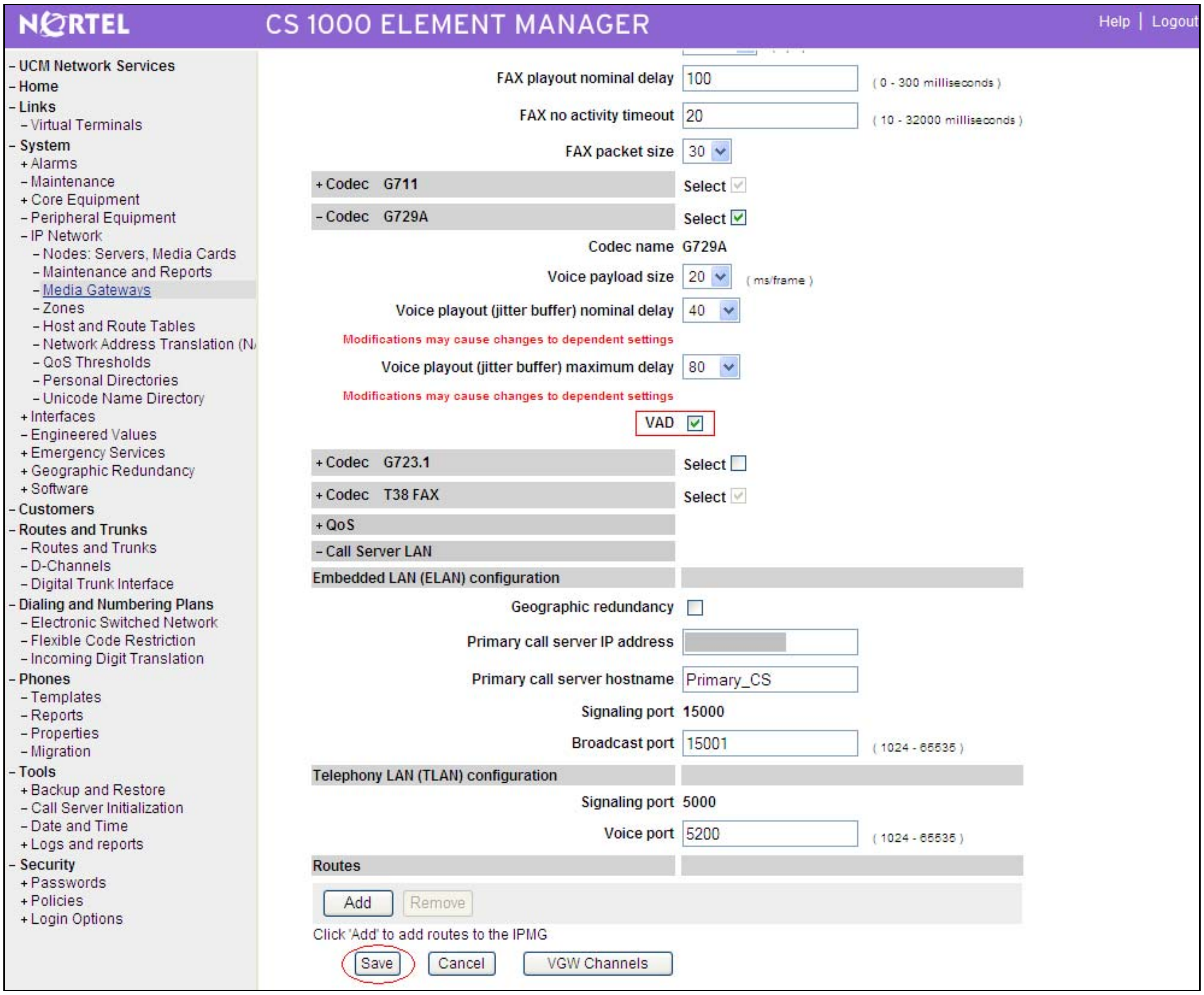
This section describes configuration of the G.729 audio codec for the Media Gateway Controller (MGC) to support calls between the B179 and non-IP telephones. On the left column menu of the main Element Manager page, navigate to **IP Network → Media Gateways**. Click on the **IPMG** that supports the digital and analog phones in the system.



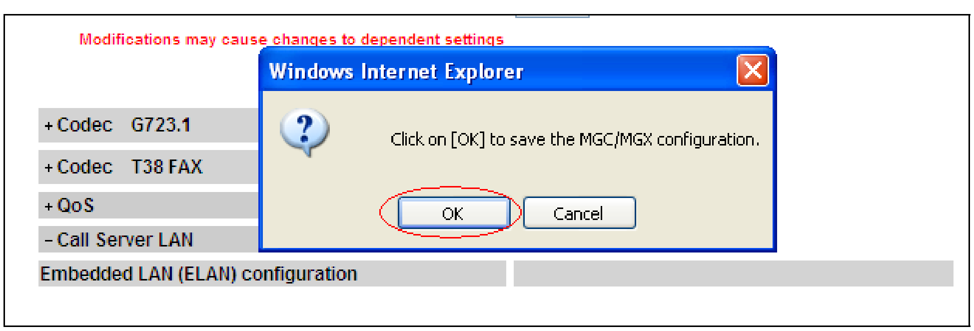
On the **IPMG Property Configuration** screen, click **Next** (not Shown). Expand the **VGW and IP phone codec profile** section. In that section, check the **Select** checkbox next to and expand the **Codec G729A** section.



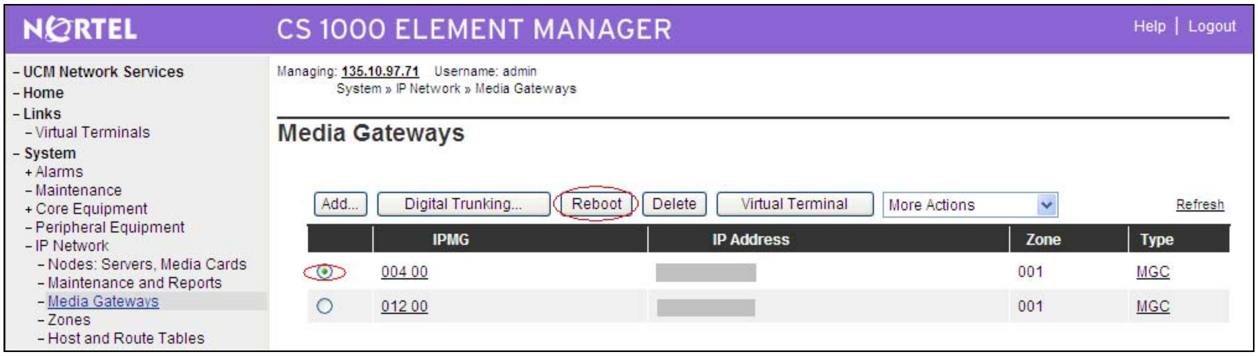
If Annex B support is desired as in the sample configuration, check the **VAD** checkbox. Note that the VAD setting should be consistent with the VAD setting in the B179 configuration. Click **Save**.



Click on **OK** to save the configuration.

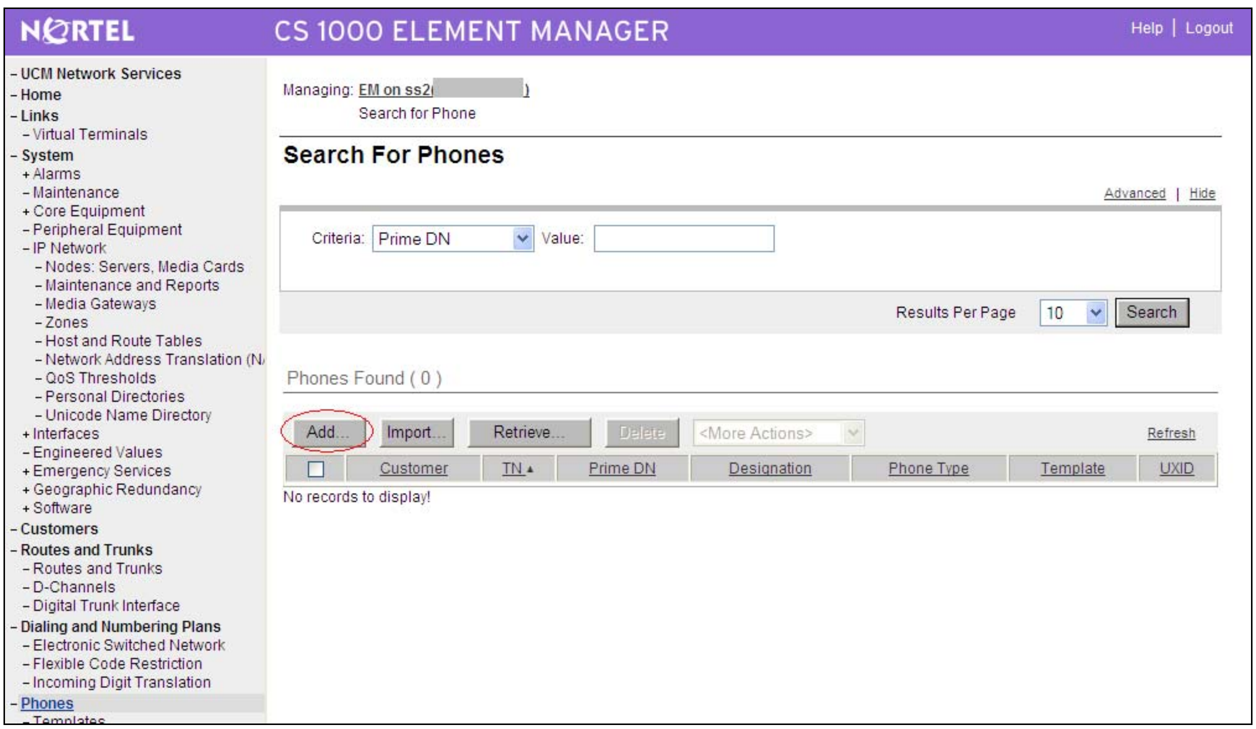


When the Media Gateway screen returns, select the radio button for the **IPMG** and click **Reboot**.

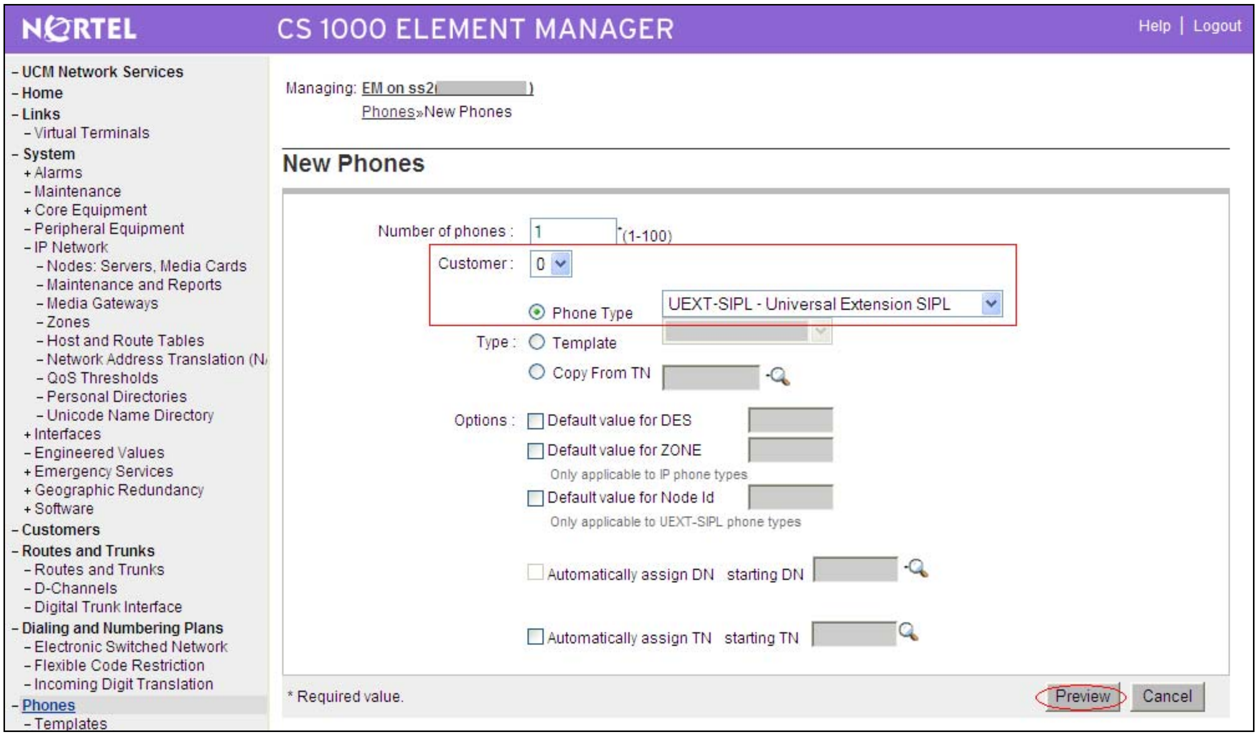


3.11. Configure SIP Line Telephone

This section describes the screens for configuring a SIP Line telephone to support the Avaya B179 SIP Conference Telephone. On the left column menu of the main Element Manager page, navigate to **Phones**. On the **Search For Phones** page, click **Add…**.

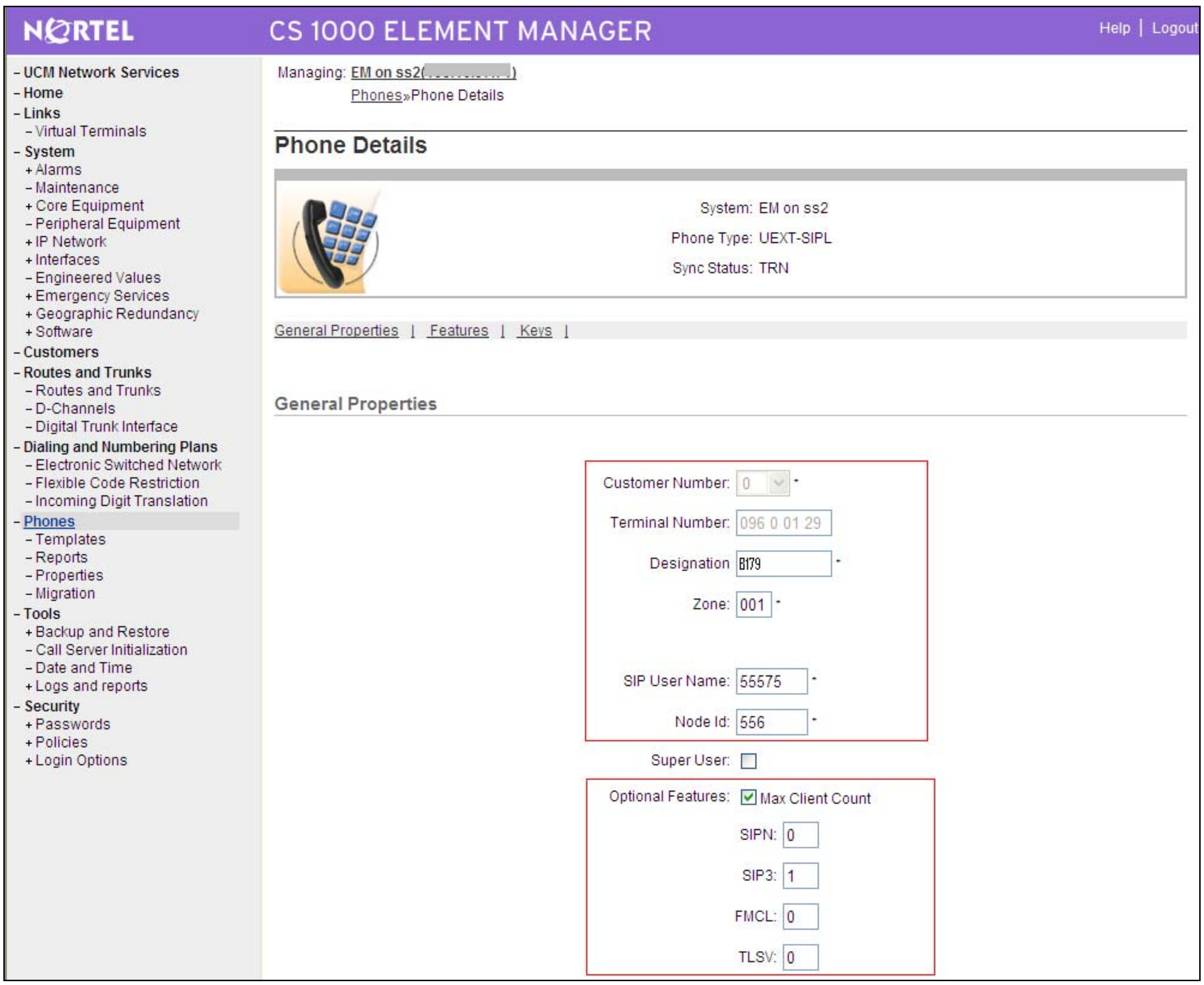


On the **New Phones** page, select the **Customer**, select the **Phone Type** radio button, and then select **UEXT-SIPL – Universal Extension SIPL**. Click **Preview**.



The following screens show the parameter values after the phone has been added. In the **General Properties** section, fill in the following fields, and leave the remaining fields at their default values:

Customer Number	Select the customer number
Terminal Number	Enter a free TN number
Designation	Enter a reference name
Zone	Enter the zone from **Section 3.7**
SIP User Name	The phone extension number used to log in at the phone
Node Id	The ID of this node
Optional Features: Max Client Count	Select the check box
SIPN	Set to **0**
SIP3	Set to **1**

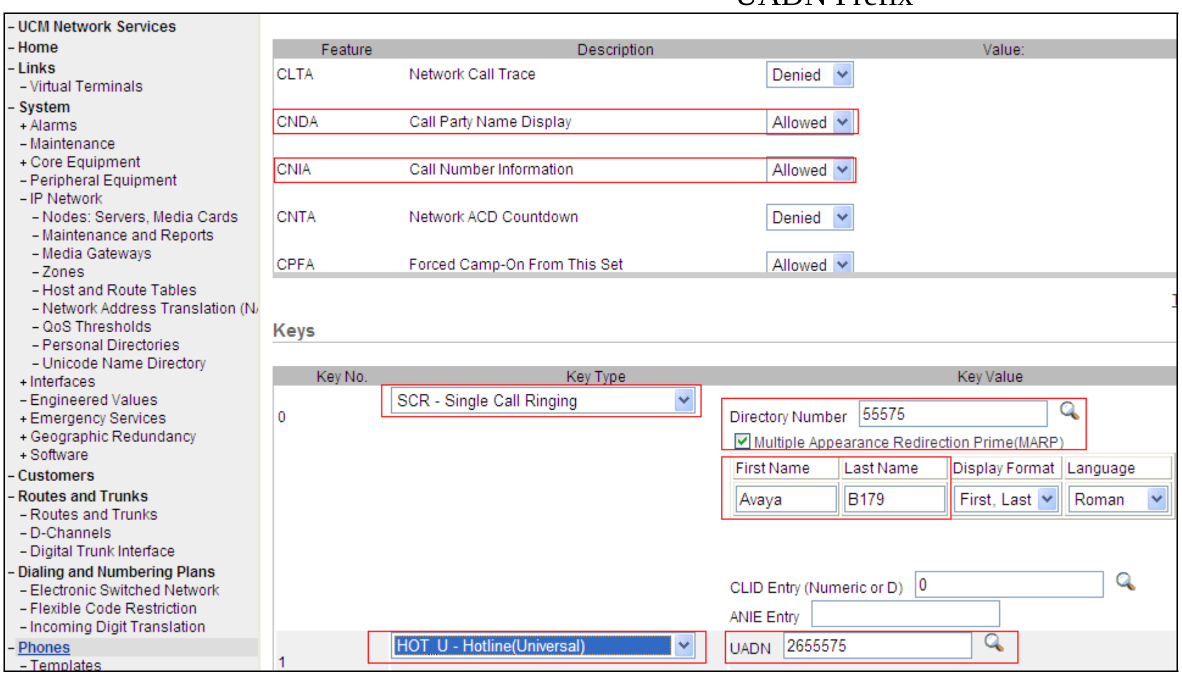


In the **Features** section, fill in the following fields, leaving the remaining fields at their defaults.

Call Party Name Display (CNDA)	**Allowed**
Call Number Information (CNIA)	**Allowed**
Restricted Conference or Transfer (FTTC))	**Unrestricted Conf. or Transfer**
Media Security Encryption (MSEC)	**Media Security Never (MSNV)**
Station Control Password (SCPW)	Enter password used to log in at the phone
Trunk Group Access Restriction (TGAR)	Set appropriately
Instrument Type (TYPE)	**UEXT**
Universal Extension User (UTXY)	**SIPL**

In the **Keys** section, fill in the following:

Key No. 0	**SCR – Single Call Ringing**
Directory Number	Phone extension number
Multiple Appearance Redirection Prime (MARP)	Select the Checkbox
First Name	Enter a name
Last Name	Enter a name
Key No. 1	**HOT_U – Hotline(Universal)**
UADN	The phone extension prefixed by the UADN Prefix[3]



Click **Save** (not shown) to save the configuration for this phone.

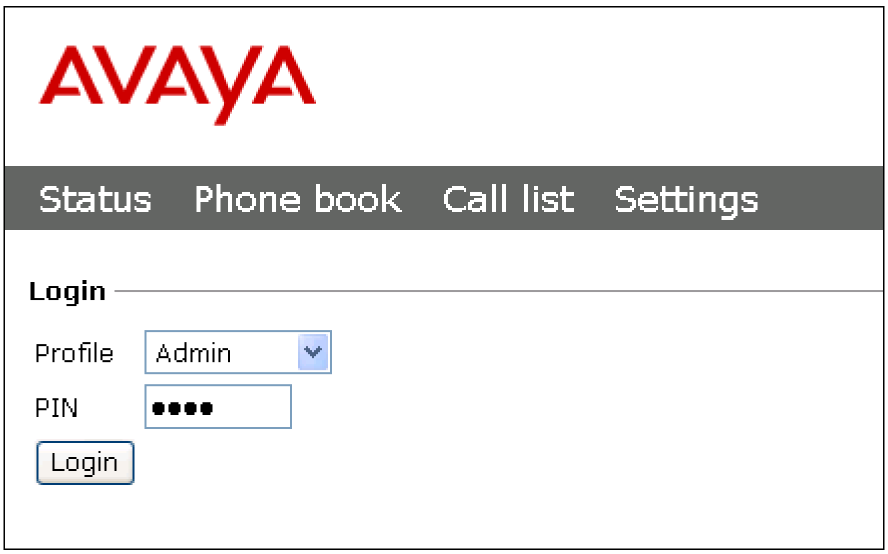
[3] The UADN is used to make and receive calls between the SIP Line Gateway and the Universal Extensions. However, this key is used only by the SIP Line Gateway (SLG) application. The UADN is not dialed by end users. It is only used internally between the Call Server and the SIP Line Gateway application. See **Section 3.2**.

4. Configure Avaya B179 IP Conference Phone

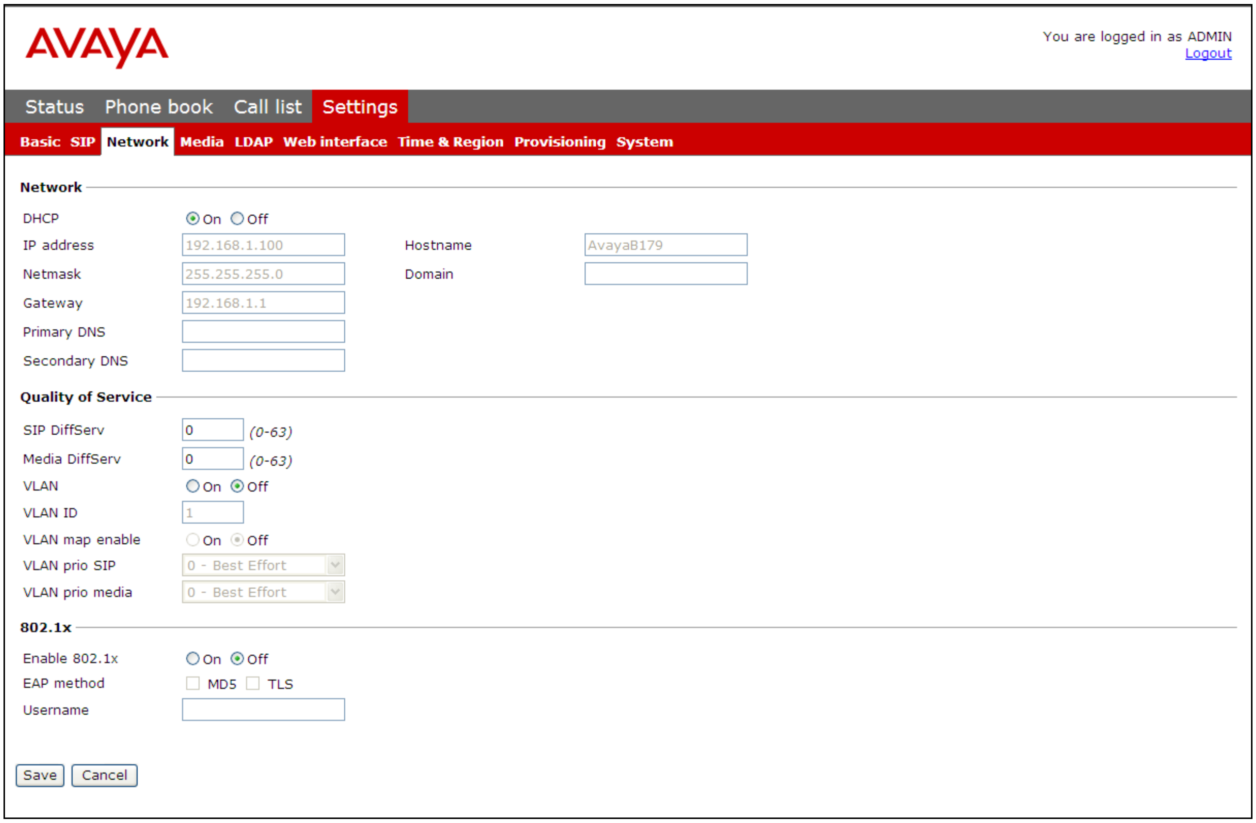
This section describes how to access the B179 web interface and configure the phone to register to Avaya Communication Server 1000. It assumes that the telephone has been administered an IP address either through DCHP or static configuration. Additional configuration details are provided in **[3]**.

4.1. SIP Registration

In the web browser address field, enter the B179 IP address. The login page will appear as shown below. Enter **admin** for the user name and the appropriate password.



Click **Login**, and the main configuration screen appears as shown on the next page, where **Settings → Network** has been selected and shows the DHCP network configuration that was configured on the B179 in the sample configuration.


AVAYA
You are logged in as ADMIN
Logout
Status
Phone book
Call list
Settings
Basic
SIP
Network
Media
LDAP
Web interface
Time & Region
Provisioning
System
Network
DHCP
On
Off
IP address
192.168.1.100
Hostname
AvayaB179
Netmask
255.255.255.0
Domain
Gateway
192.168.1.1
Primary DNS
Secondary DNS
Quality of Service
SIP DiffServ
0
(0-63)
Media DiffServ
0
(0-63)
VLAN
On
Off
VLAN ID
1
VLAN map enable
On
Off
VLAN prio SIP
0 - Best Effort
VLAN prio media
0 - Best Effort
802.1x
Enable 802.1x
On
Off
EAP method
MD5
TLS
Username
Save
Cancel

4.2. Configure SIP Signalling Settings

To configure the SIP signalling settings, navigate to **Settings → SIP**, and fill in the following:

<u>Under **Account 1**</u>:

Enable account	Select the **Yes** radio button
Account name	Meaningful name for account status display on phone screen
User	Extension (**SIP User Name**) of the SIP Line telephone configured in **Section 3.11**
Realm	Use the default of "*"
Registrar and **Proxy**	SIP domain configured in the CS 1000, with Port number
Authentication name	Extension (**SIP User Name**) of the SIP Line telephone configured in **Section 3.11**
Password	The **Station Control Password** of the SIP Line telephone configured in **Section 3.11**
Registration interval	Enter a value (300 was used in the sample configuration)

<u>Under **Advanced**</u>:

Enable blind transfer	Select the **No** radio button[4]
Outbound proxy	No input was used in this configuration

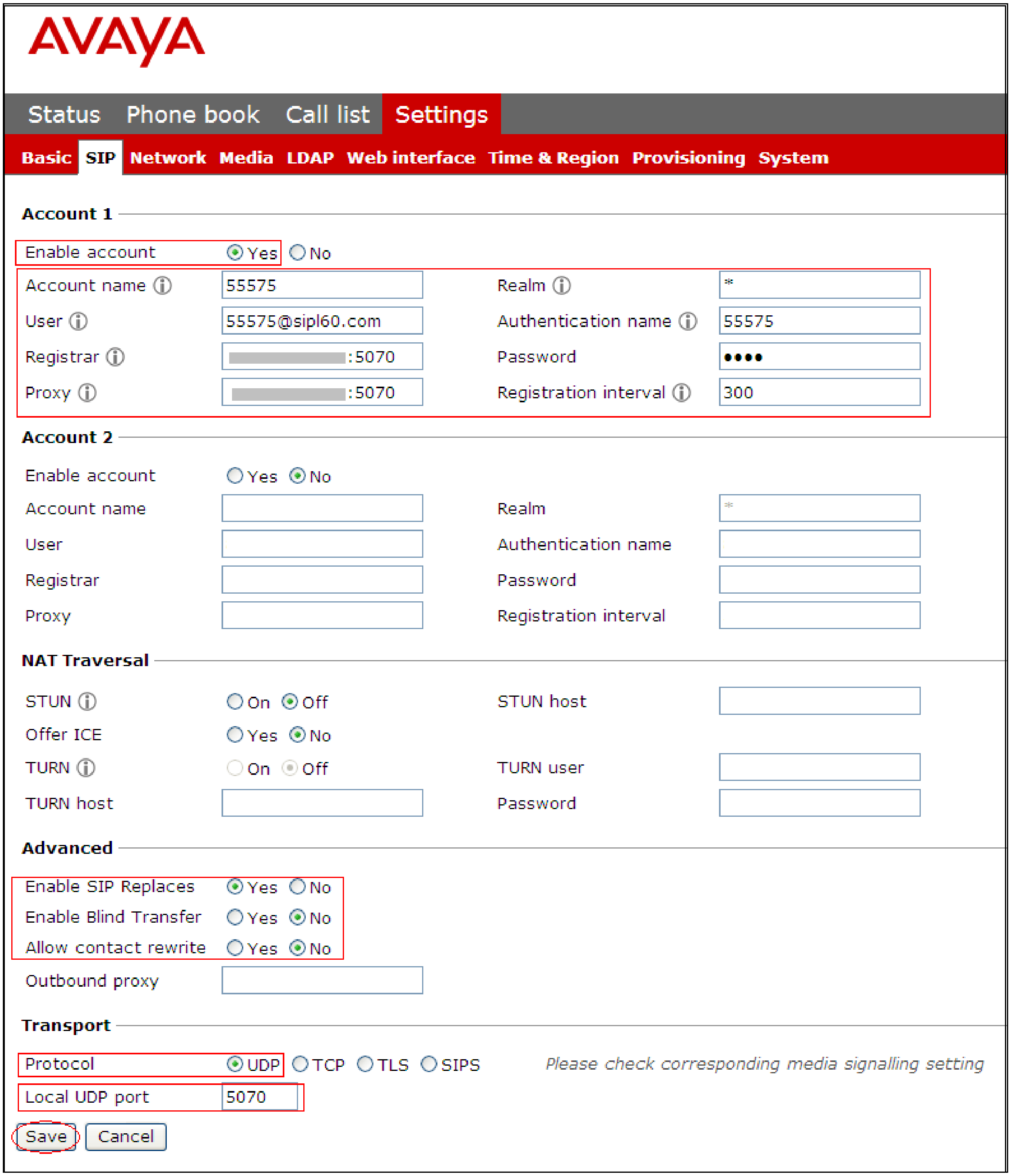
<u>Under **Transport**</u>:

Protocol	Select the **TCP** or **UDP** radio button (UDP shown)
Local Port	Enter **5070**

Click **Save**.

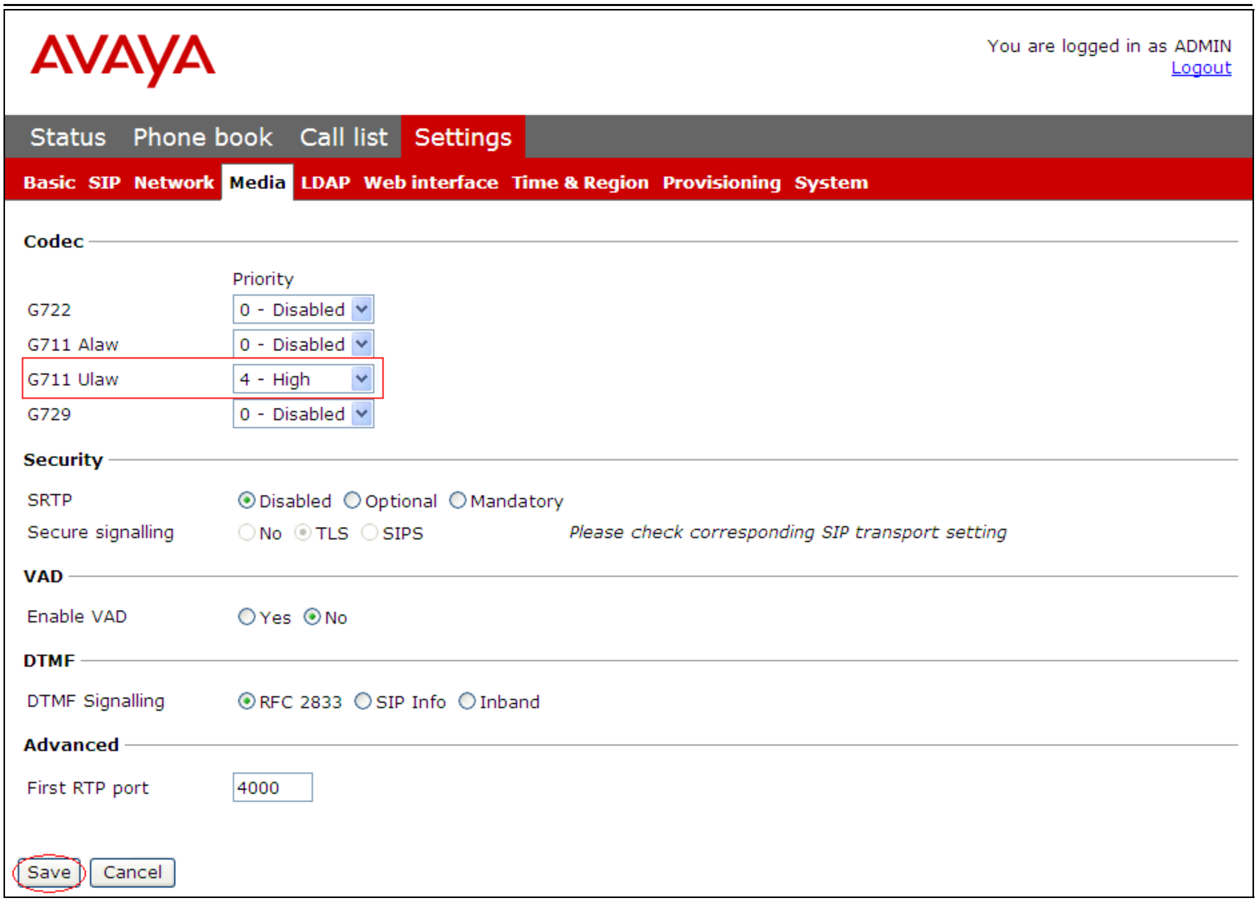
The SIP configuration screen is shown on the next page.

[4] This feature is not yet supported in this configuration

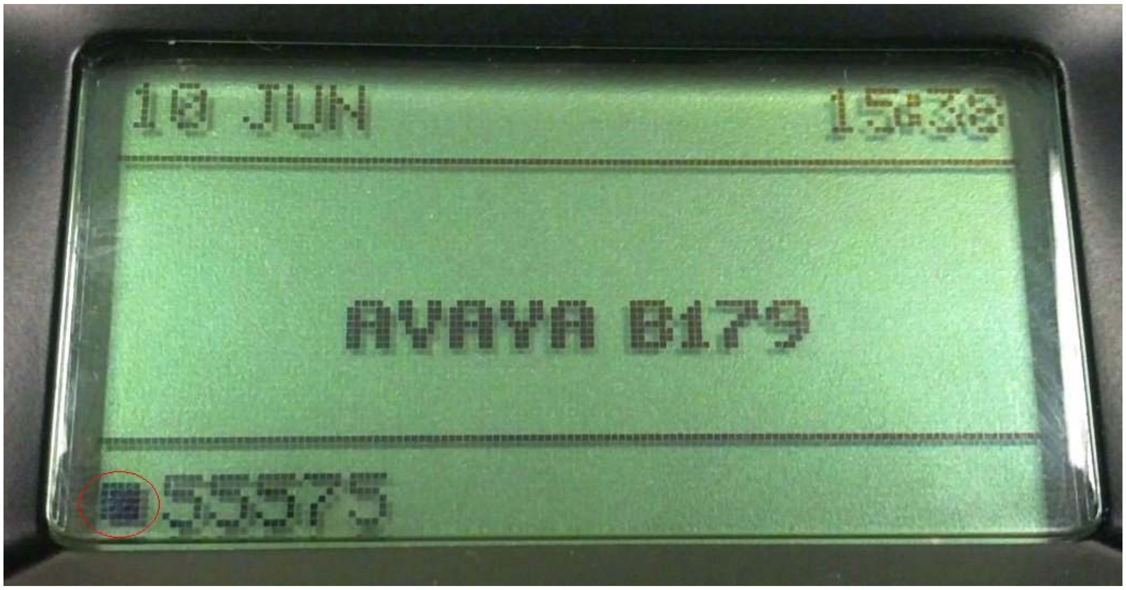

AVAYA
Status
Phone book
Call list
Settings
Basic
SIP
Network
Media
LDAP
Web interface
Time & Region
Provisioning
System
Account 1
Enable account
Yes
No
Account name
55575
Realm
*
User
55575@sipl60.com
Authentication name
55575
Registrar
:5070
Password
••••
Proxy
:5070
Registration interval
300
Account 2
Enable account
Yes
No
Account name
Realm
*
User
Authentication name
Registrar
Password
Proxy
Registration interval
NAT Traversal
STUN
On
Off
STUN host
Offer ICE
Yes
No
TURN
On
Off
TURN user
TURN host
Password
Advanced
Enable SIP Replaces
Yes
No
Enable Blind Transfer
Yes
No
Allow contact rewrite
Yes
No
Outbound proxy
Transport
Protocol
UDP
TCP
TLS
SIPS
Please check corresponding media signalling setting
Local UDP port
5070
Save
Cancel

4.3. Media Configuration

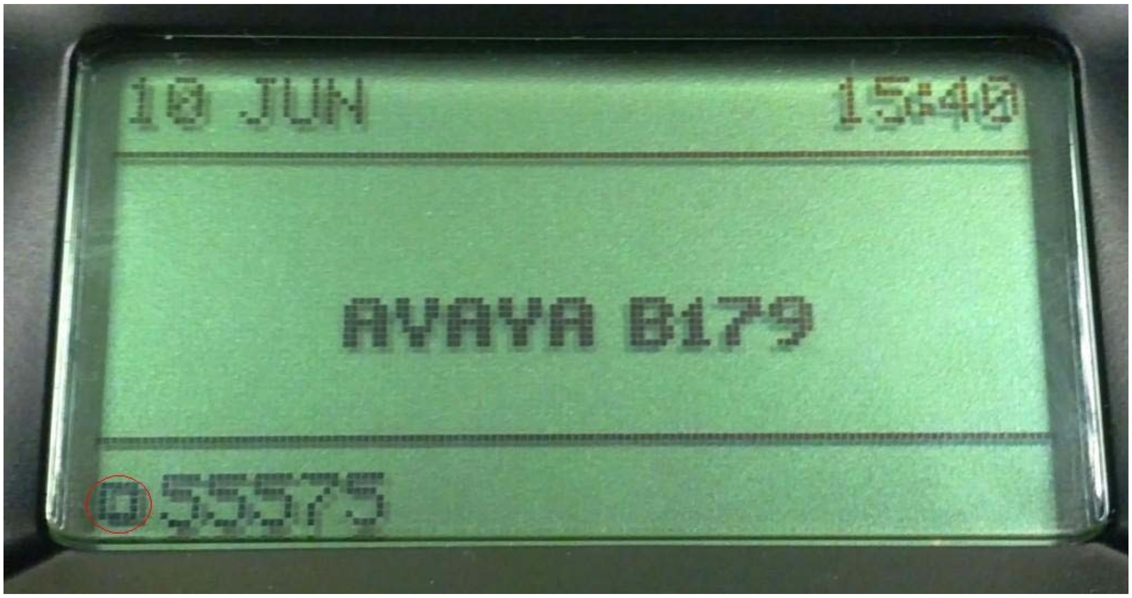
To configure the audio codec settings, navigate to **Settings → Media**, and select the priority for codec selection. CS1000 6.0 does not support the G7.22 Codec, so this selection should be set to **0 – Disabled**. Because there were some interoperability issues with G.729 it is recommended that this option should also be set to **0 – Disabled.** Set the G711 Ulaw codec to **4 – High.** Defaults can be used for the remaining fields. Click on **Save** when done.



After the configuration has been saved, the B179 will register with the CS 1000, and a display similar to those shown in the figures below will appear on the telephone.   The **Hostname** is displayed at the center, and in the lower left corner is the **Account name**. To the left of the **Account name** is a square icon that indicates the SIP registration status of the B179. If the square is filled in as shown below, the B179 has successfully registered.



If the square is not filled in, registration was unsuccessful.



5. Observations

During testing with this configuration the following observations were made:

- Calls from the B179 via the inbound call log are not supported.
- Group conference by the B179 is not supported.
- The use of the G.729audio codec is not recommended because of possible interoperability issues.
- It is recommended to disable Blind Transfer as indicated in section 4.2 because of possible interoperability issues.

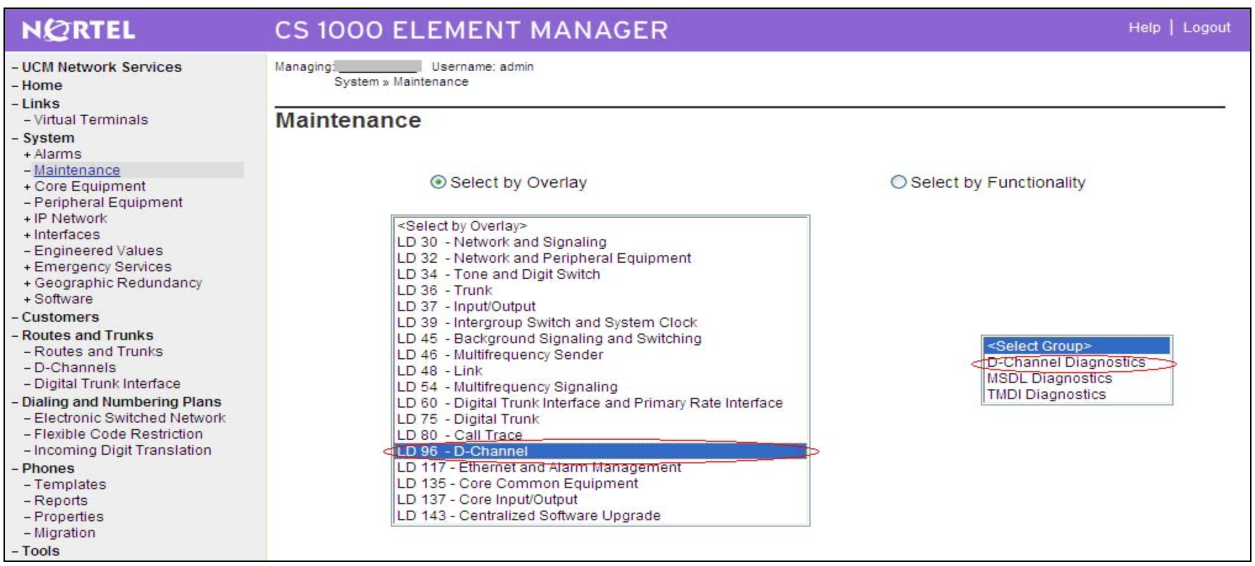
6. Verification Steps

This section provides tests that can be performed to verify proper configuration of the CS 1000 and B179.

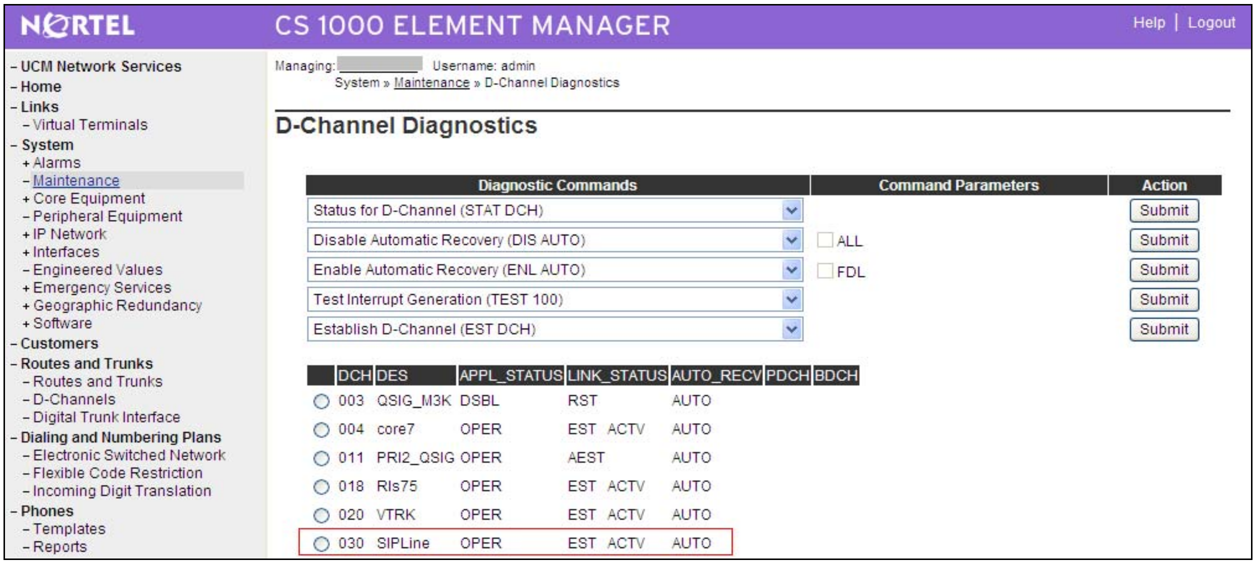
6.1. Verify Avaya Communication Server 1000

6.1.1. Verify D-Channel Status

Verify status of the SIP Line D-Channels by navigating to **System → Maintenance**, selecting **Select by Overlay**, **LD 96 – D-Channel**, and **D-Channel Diagnostics**.

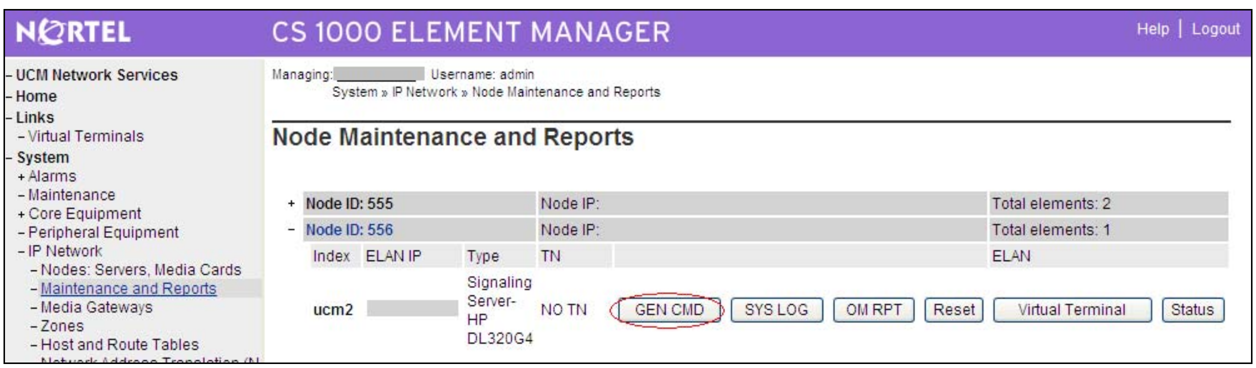


The screen below shows the **APPL_STATUS** of the SIP Line D-Channel as "OPER" and the **LINK_STATUS** as "EST ACTV". This is normal.

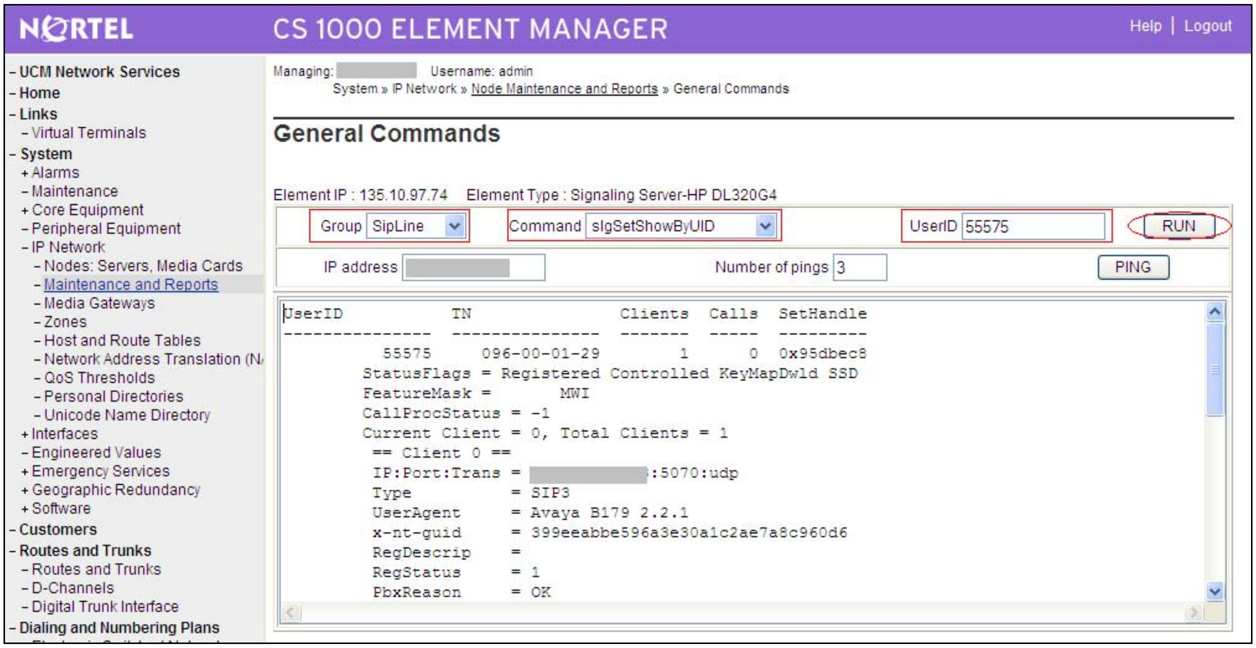


6.1.2. Verify SIP Registration Status

In the Element Manager Web interface, navigate to **System → IP Network → Maintenance and Reports** on the left pane. If there are multiple Nodes, select the node of the SIP Line. Click **GEN CMD**.

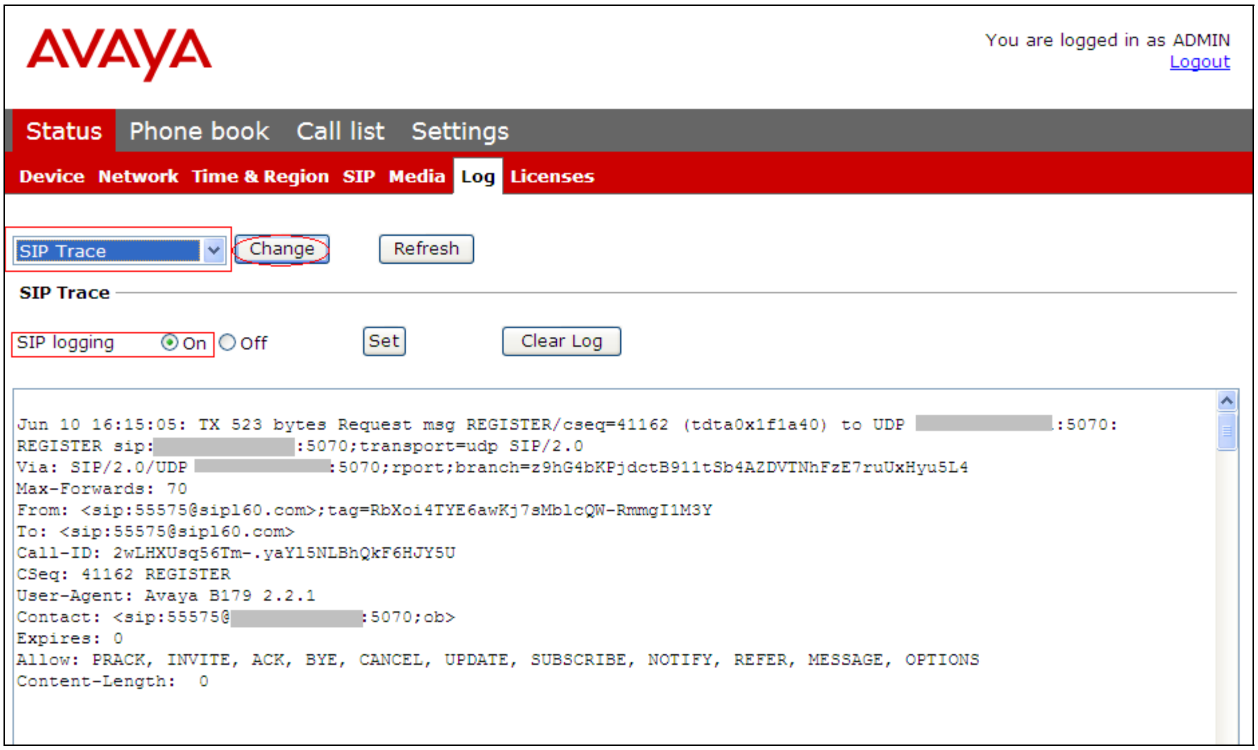


The **General Commands** page is displayed. From the **Group** drop-down menu select **SipLine**, from the **Command** drop-down menu select **slgSetShowByUID**, enter the B179 extension in **UserID**, and click on **RUN**. The output shown indicates successful registration and displays details of the registration parameters. Note that if the B179 has not registered, the error message "Invalid userId 55575" will be returned instead of the detailed registration information.


NORTEL
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 (N
- 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
Managing: Username: admin
System » IP Network » Node Maintenance and Reports » General Commands
General Commands
Element IP : 135.10.97.74 Element Type : Signaling Server-HP DL320G4
Group SipLine
Command slgSetShowByUID
UserID 55575
RUN
IP address
Number of pings 3
PING
UserID TN Clients Calls SetHandle
55575 096-00-01-29 1 0 0x95dbec8
StatusFlags = Registered Controlled KeyMapDwld SSD
FeatureMask = MWI
CallProcStatus = -1
Current Client = 0, Total Clients = 1
== Client 0 ==
IP:Port:Trans = :5070:udp
Type = SIP3
UserAgent = Avaya B179 2.2.1
x-nt-guid = 399eeabbe596a3e30a1c2ae7a8c960d6
RegDescrip =
RegStatus = 1
PbxReason = OK

6.2. Verify Avaya B179 SIP Conference Phone

Successful registration of the phone can be verified by inspecting the status icon to the left of the **Account name**, shown at the lower left of the telephone display as described in Section 4.3. Registration and call tracing can be performed on the B179 by navigating to **Status** → **Log**. Select **SIP Trace** on the left and click **Change**. Ensure that the **SIP logging** radio button is selected. After attempting registration, click **Refresh** to see the result. The log can be cleared at any time by clicking **Clear Log**. The screen below shows the REGISTER message sent by the B179 for a successful registration to the CS 1000.



7. Conclusion

As illustrated in these Application Notes, Avaya Communication Server 1000 Release 6.0 and the Avaya B179 SIP Conference Phone can be used together in an integrated solution.

8. Additional References

Product documentation for Avaya products may be found at http://support.avaya.com.

[1] *Communication Server 1000 - Element Manager System Reference – Administration*, Release: 6.0, Document Revision: 03.20, Document #NN43001-632.

[2] *Communication Server 1000 SIP Line Fundamentals*, Release 6.0, Document #NN43001-508, 01.08, 9 February 2010.

[3] *Avaya B179 SIP Conference Phone Installation and Administration Guide*

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com