Avaya Solution & Interoperability Test Lab

Application Notes for Avaya B179 SIP Conference Phone with Avaya Communication Server 1000 Release 7.5 – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Server 1000 Release 7.5 and the Avaya B179 SIP Conference Phone. The B179 is a SIP VoIP conference telephone that registers as a standard SIP Line client with Communication Server 1000. 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 at the request of Product Management.

1. Introduction

These Application Notes describe a solution comprised of Avaya Communication Server 1000 Release 7.5 and the Avaya B179 SIP Conference Phone. The B179 is a SIP VoIP conference telephone that registers as a standard SIP Line client with Communication Server 1000. 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 at the request of Product Management.

As shown in **Figure 1**, all telephones, including the B179, are registered to Avaya Communication Server 1000, which is configured as a co-resident single server system. The telephones are configured in the 57xxx extension range.

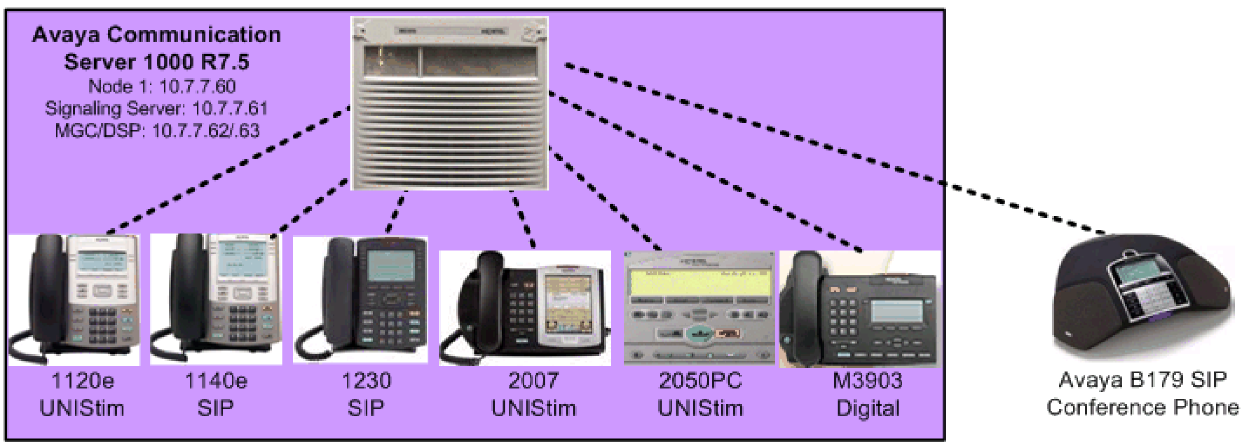


Figure 1: Network Configuration

2. Equipment and Software Validated

Provider	Hardware Component	Software Version
Avaya	Avaya Communication Server 1000E	7.50Q 7.50.17 (see **Table 2** for applied updates)
Avaya	Avaya 1120E IP Deskphone	UNIStim: 0624C8A
Avaya	Avaya 1165e IP Deskphone	SIP: 04.00.04.00
Avaya	Avaya 1230 IP Deskphone	SIP: 04.00.04.00
Avaya	Avaya 2007 IP Deskphone	UNIStim: 0621C8A
Avaya	Avaya IP Softphone 2050PC	UNIStim: 4.01.041
Avaya	Avaya M3903 Digital Phone	N/A
Avaya	B179 SIP Conference Phone	2..2

Table 1: Hardware Components and Software Versions

Update Type	Update Components
Patch	None
Service Pack	cs1000-baseWeb-7.50.17.01-1.i386.000
	cs1000-dbcom-7.50.17-01.i386.000
	cs1000-sps-7.50.17-01.i386.000
	cs1000-linuxbase-7.50.17.04-00.i386.000
	cs1000-Jboss-Quantum-7.50.17.01-1.i386.000
	cs1000-bcc-7.50.17.03-00.i386.000
	cs1000-dmWeb-7.50.17.04-00.i386.001
	cs1000-shared-pbx-7.50.17-01.i386.000
	cs1000-vtrk-7.50.17-11.i386.000
Deplist	p30588_1, p30550_1, p30613_1, p30618_1, p30621_1, p30565_1, p30597_1, p30595_1, p30591_1, p30560_1, p30594_1, p30619_1
Loadware	IPMG TYPE CSP/SW MSP APP FPGA BOOT DBL1 DBL2 4 0 MGC BD01 AB01 BA07 AA18 BA07 DSP1AB03 N/A

Table 2: CS1000E Applied Updates

3. Configure Avaya Communication Server 1000

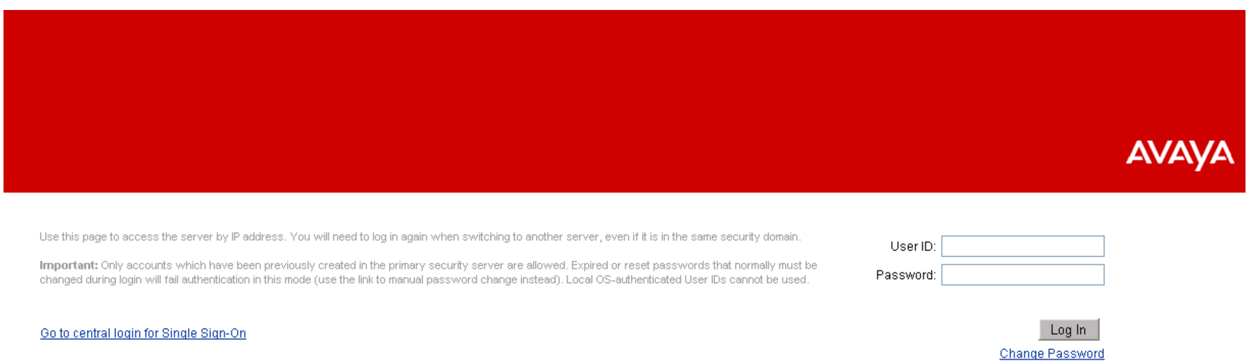
This section describes the steps to configure the following, using CS 1000 Element Manager:

- 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

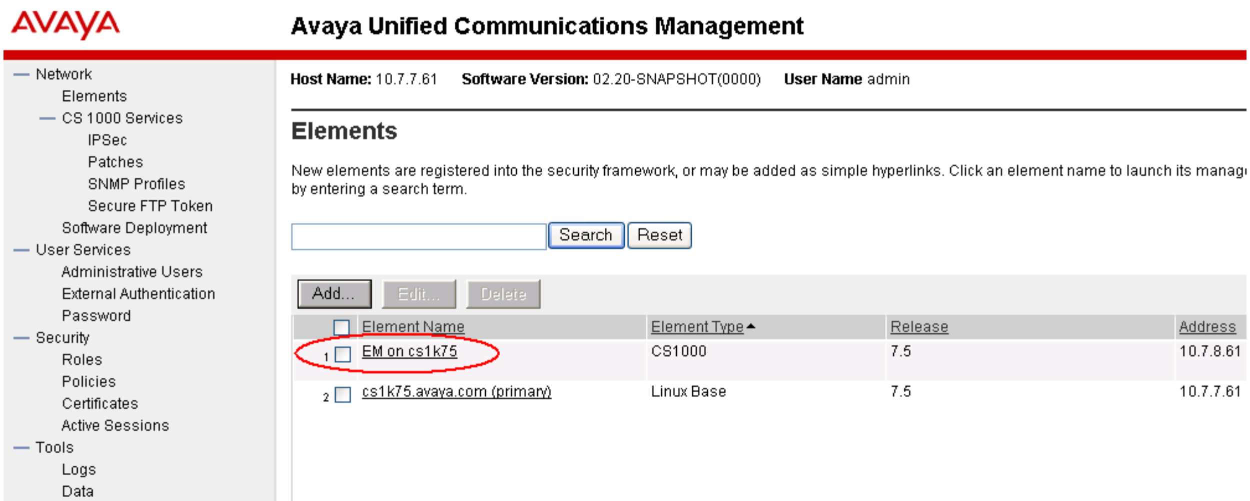
It is assumed that basic installation and configuration of the CS 1000 call server, signaling server, and node have been completed. Additional configuration details are provided in **[1, 2]**.

3.1. Log in to Element Manager (EM)

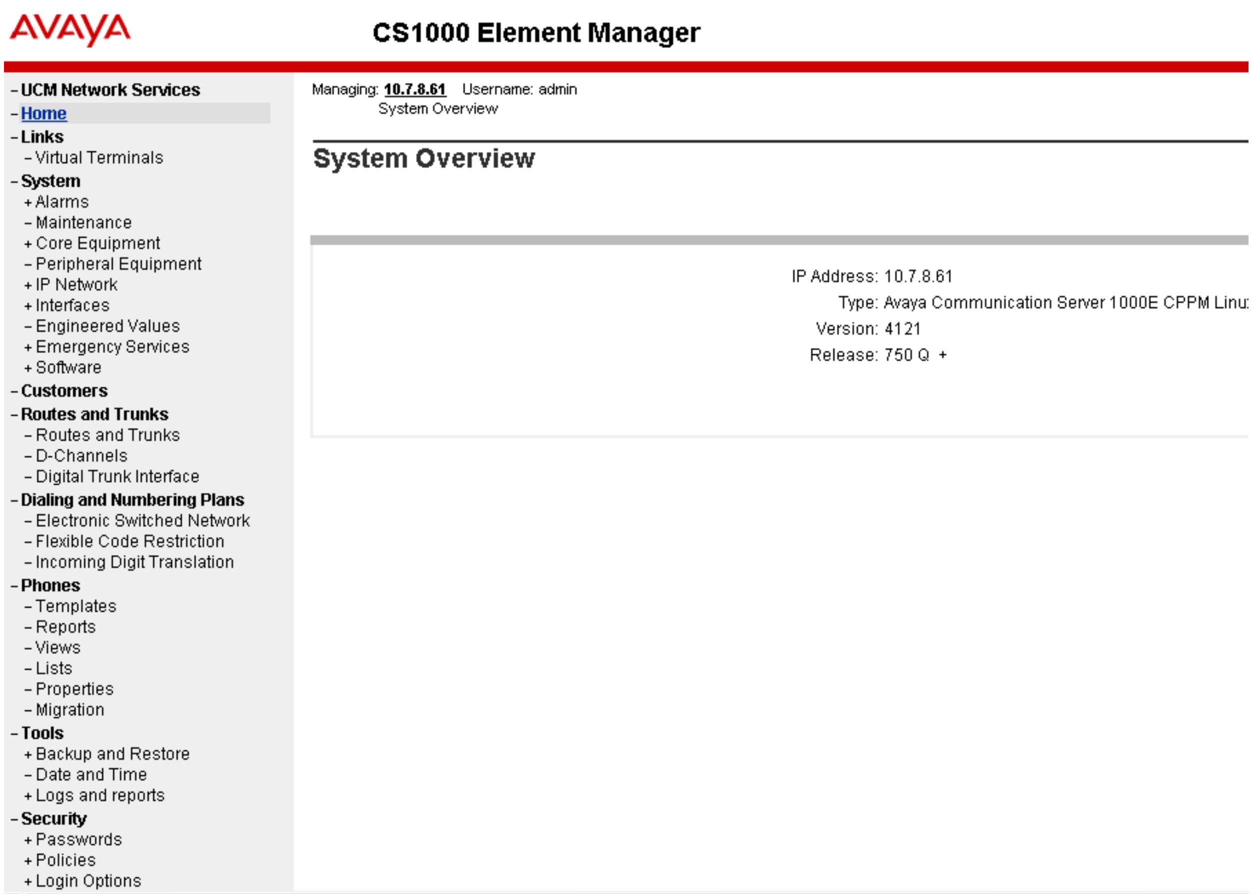
Access the Unified Communications Management (UCM) web based interface by using the URL "http://<ip-address>" in an Internet browser window, where "<ip-address>" is the IP address of the call server. Note that the IP address for the Call Server may vary, and in this case "10.7.7.61" is used. Log in with the appropriate user ID and password.



The following **Unified Communications Management** screen will be displayed. Click on the **Element Name** corresponding to the **Element Type** of "CS1000".

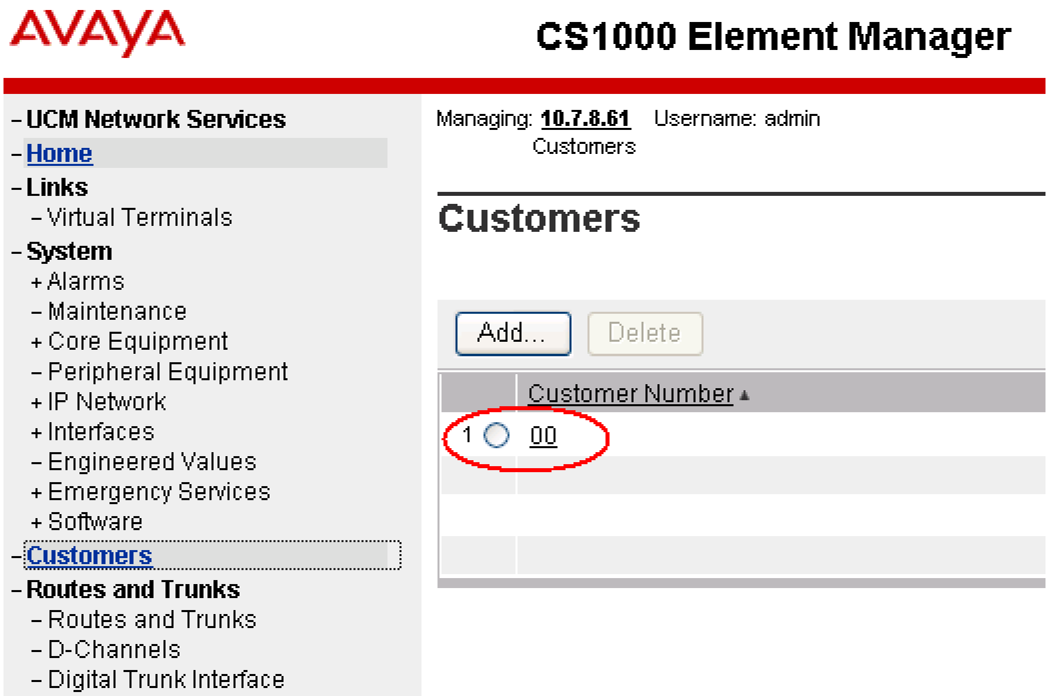


The CS 1000 Element Manager page appears as shown below.

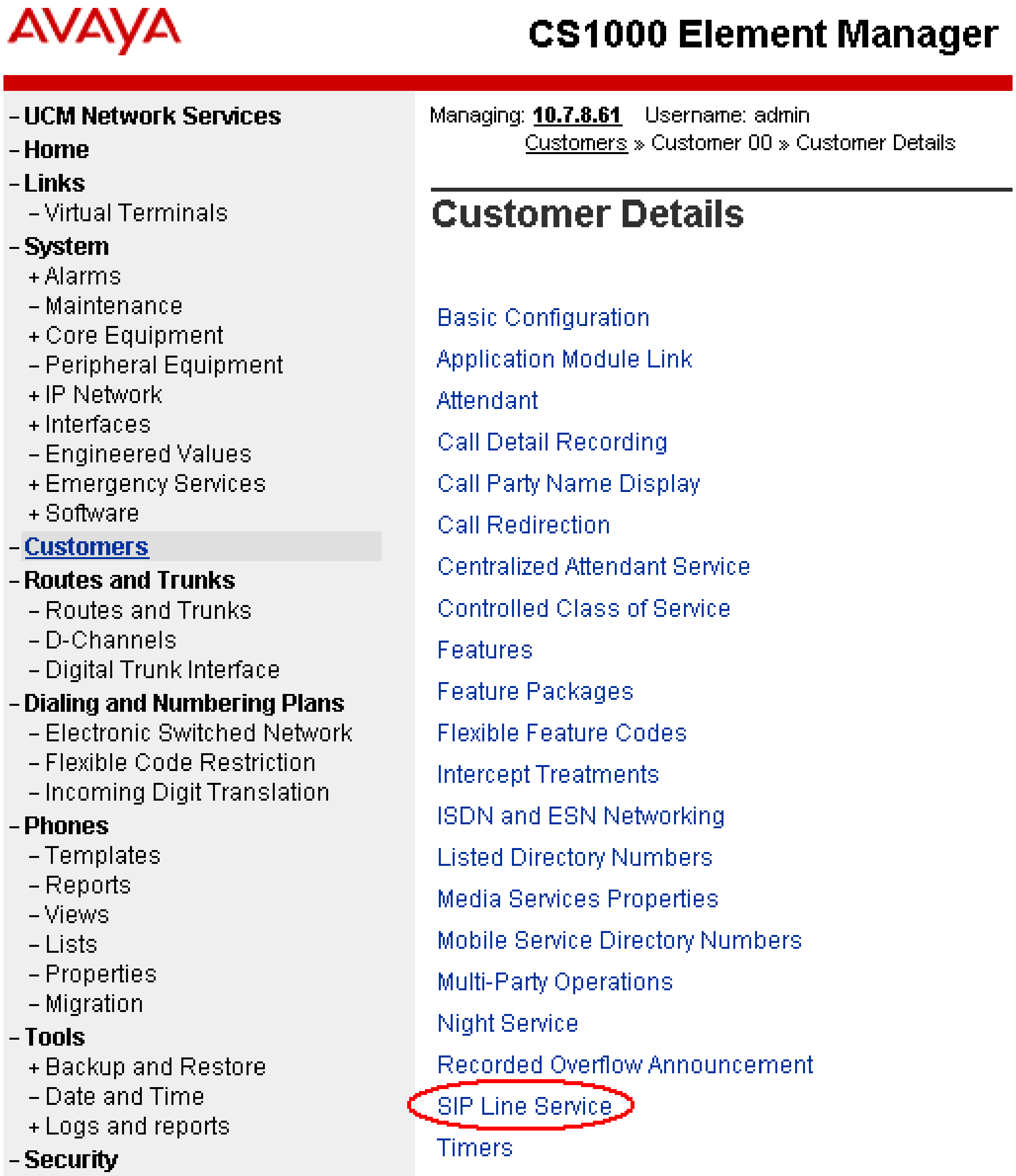


3.2. Enable SIP Line Service

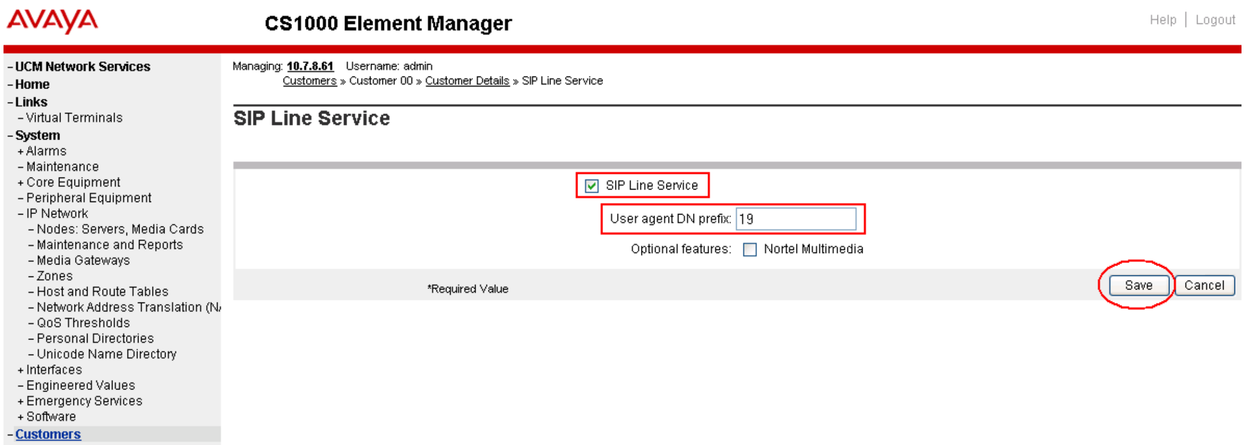
Select **Customers** in the left pane. The **Customers** screen is displayed. Click the link associated with the appropriate customer, in this case **00**. The system can support more than one customer with different network settings and options. In the sample configuration, only one customer was configured 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 **User Agent DN prefix**[1], 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 Node, as it will be used in configuring the B179 later. Select the **Node ID** on which SIP Line service is to be enabled.

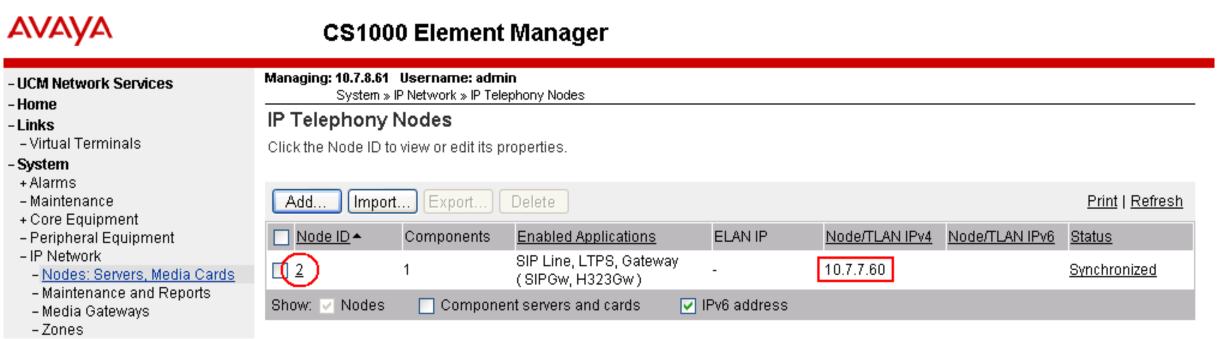


Figure 2: CS 1000 Node Screen

[1] The User Agent DN Prefix is used to form the User Agent DN. See **Section 3.11**.

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

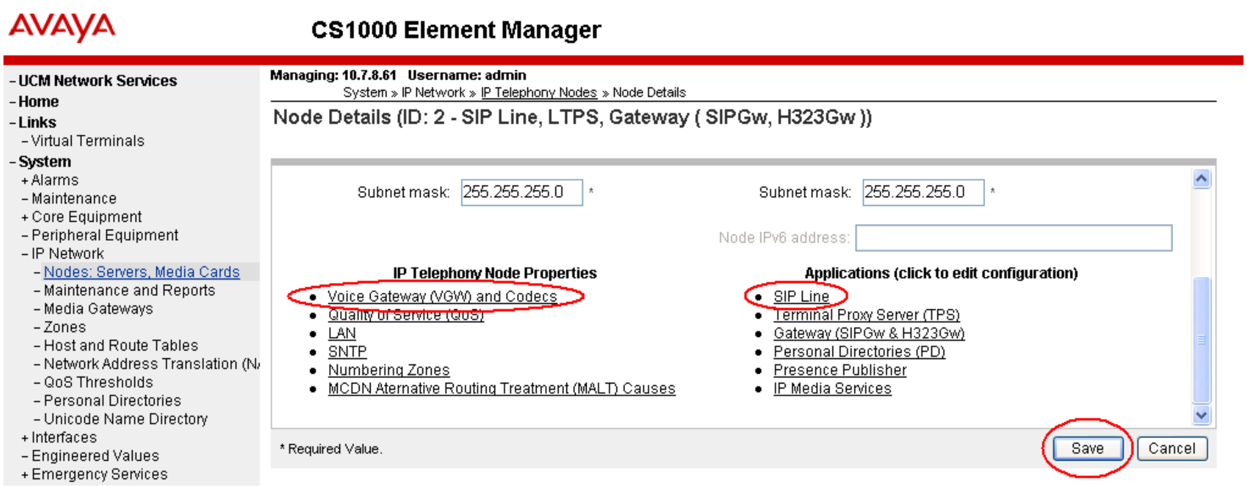
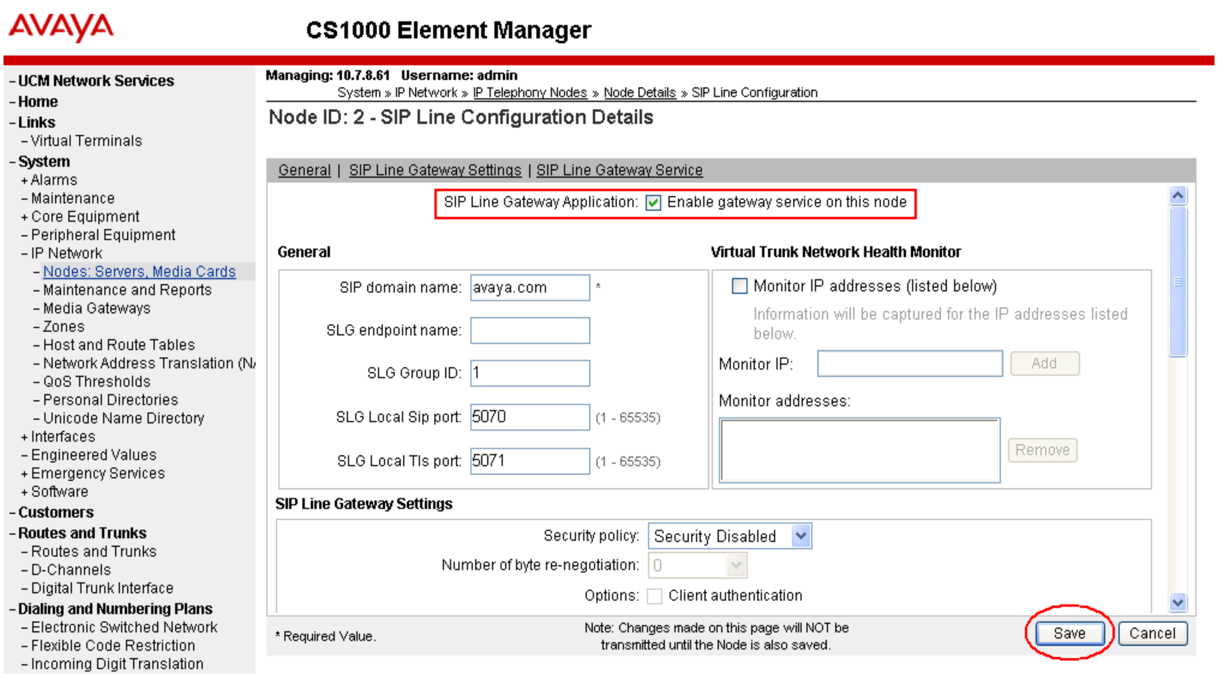


Figure 3: Node Details Screen

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



Return to the Node Details Screen (**Figure 3**) and click on **Voice Gateway (VGW) and Codecs**. For G.722 and G.729 support, check **Enabled** next to **Codec G.722:** and **Codec G.729:**. If G.729 Annex B (silence suppression) is desired as in the sample configuration, check the **Voice Activity Detection (VAD)** checkbox. Note that the VAD setting should be consistent with the VAD setting in the B179 configuration (see **Section 4.2 Figure 9**). Click **Save**. Then click **Save** on the **Node Details** screen (**Figure 3**).

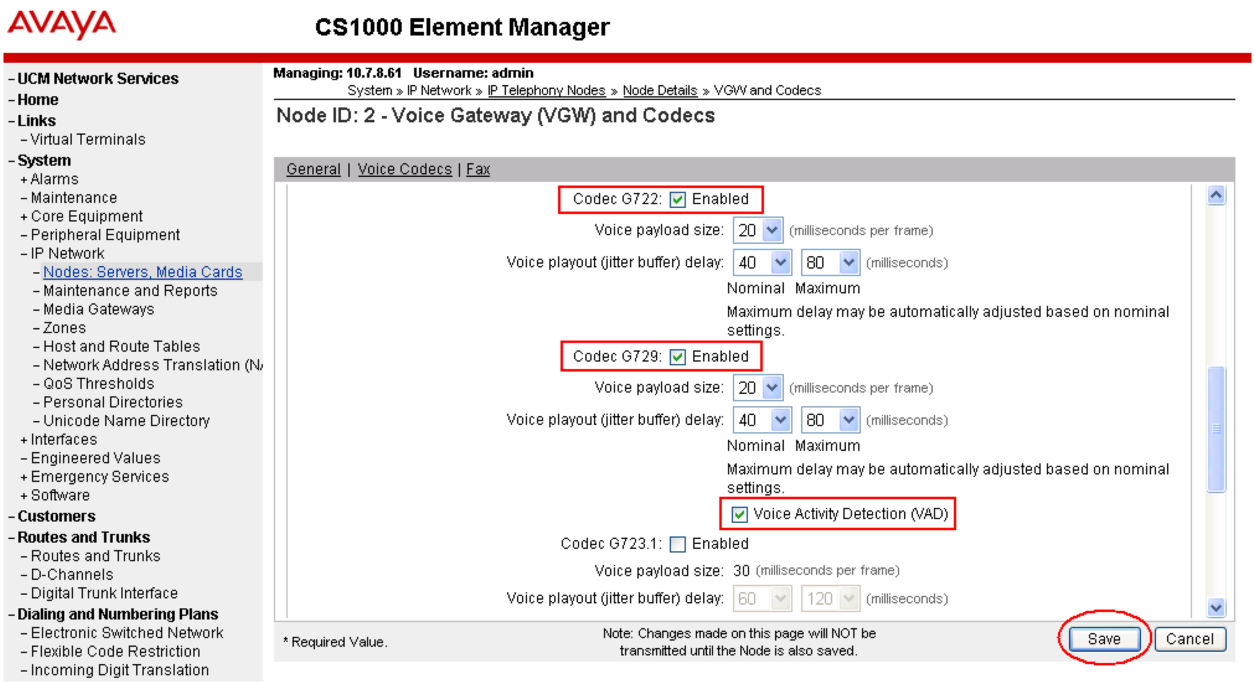
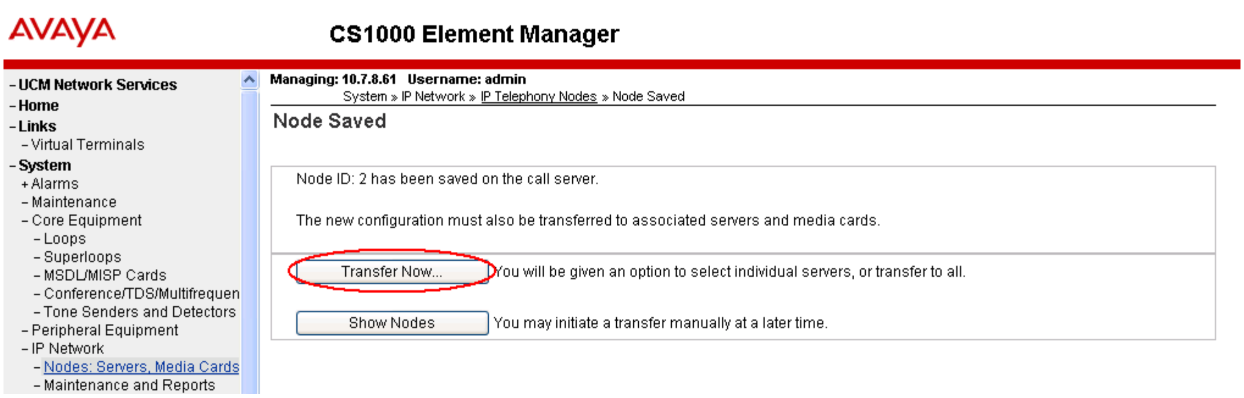
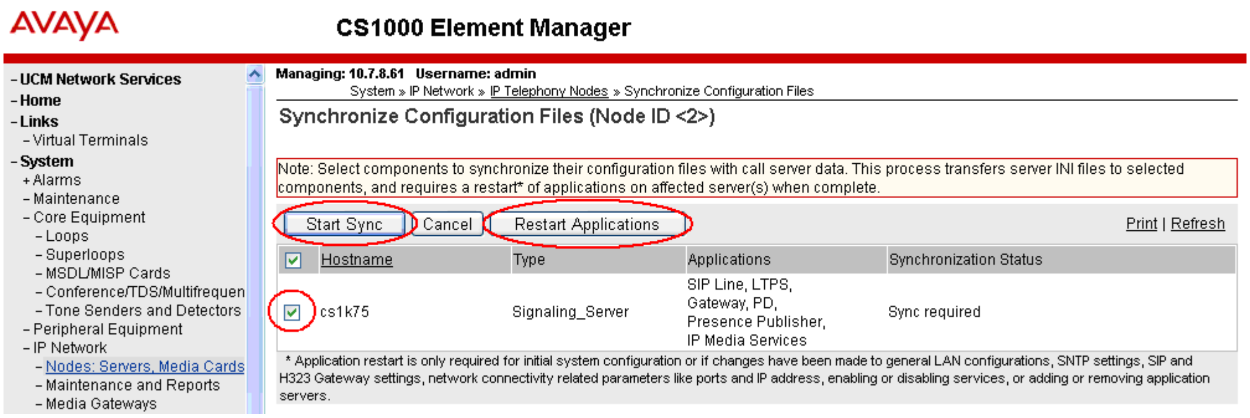


Figure 4 – Node Codec Selection

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



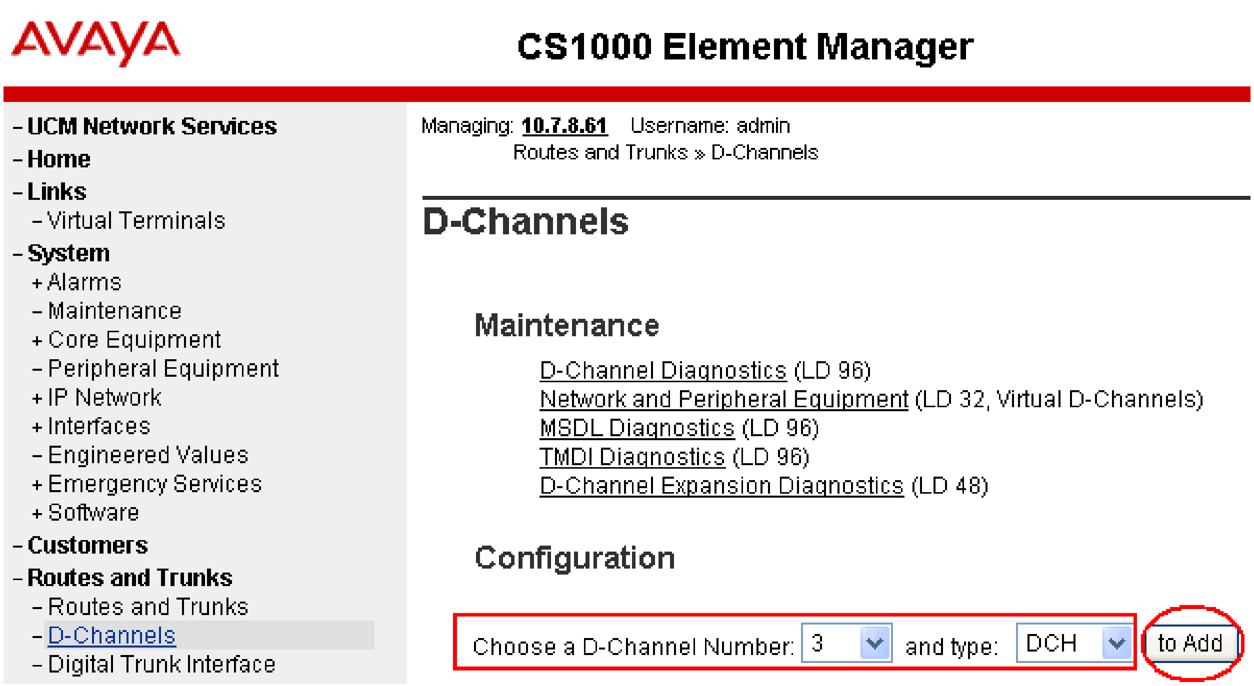
Once the transfer completes, the **Synchronize Configuration Files (Node ID <id>)** page is displayed.



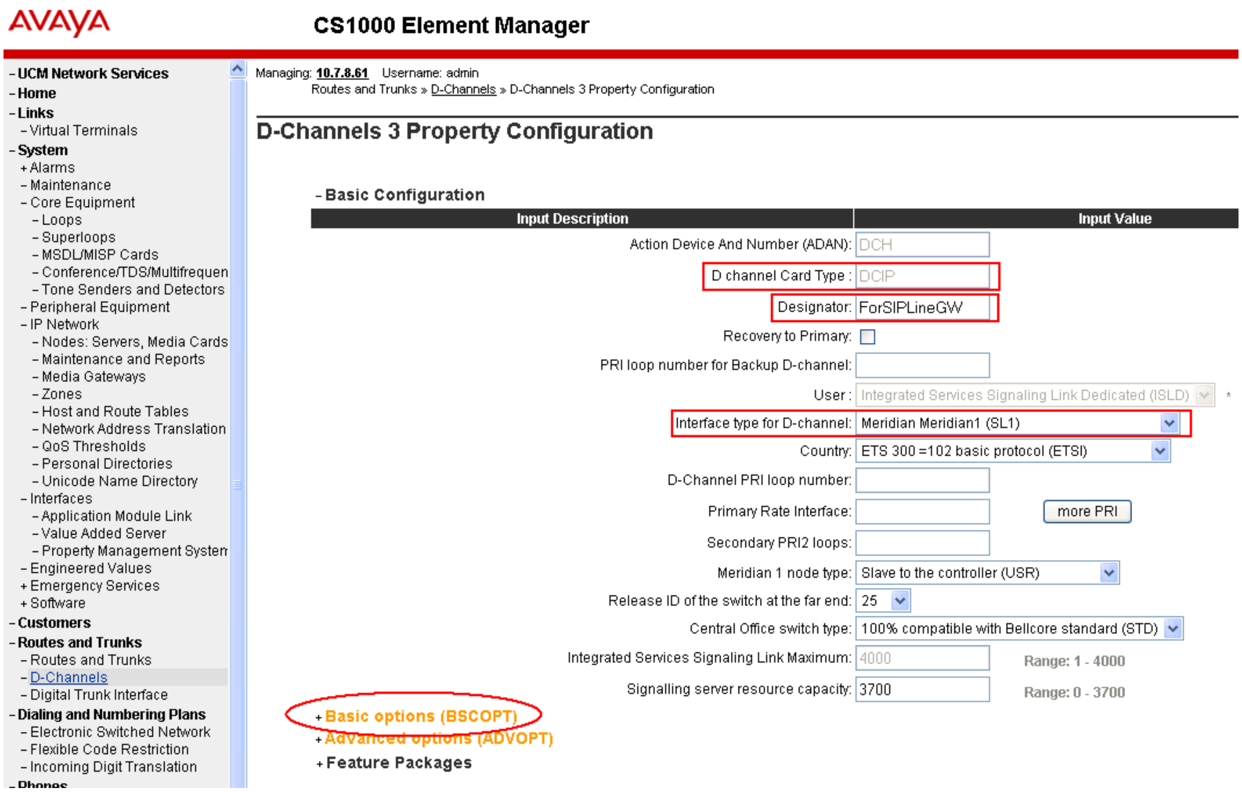
Check the appropriate Call Server and click **Start Sync.** The screen will automatically refresh until the synchronization is finished. The **Synchronization Status** field will update from **Sync required** (as shown) to **Synchronized** (not shown). After synchronization completes, click **Restart 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 3 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.**

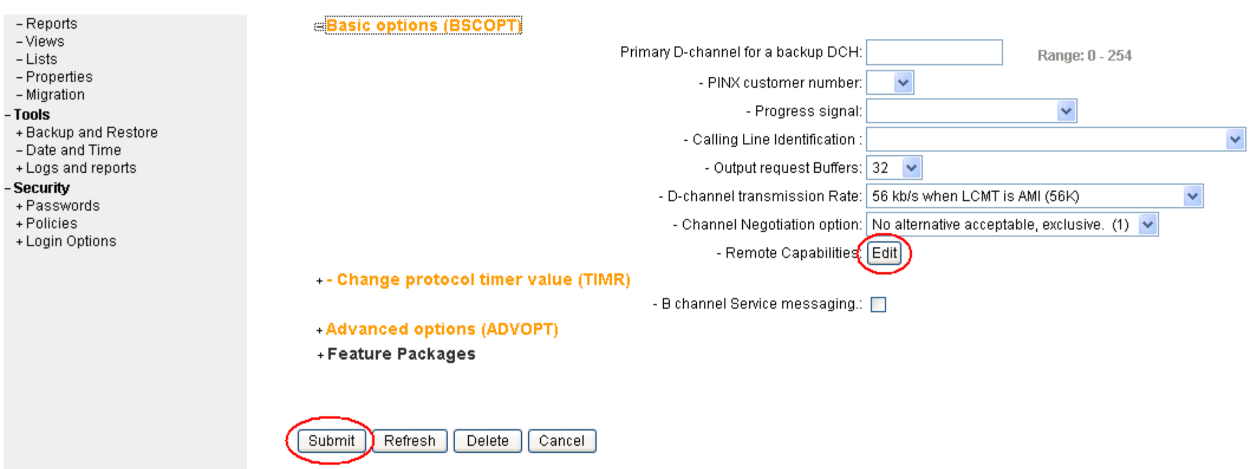
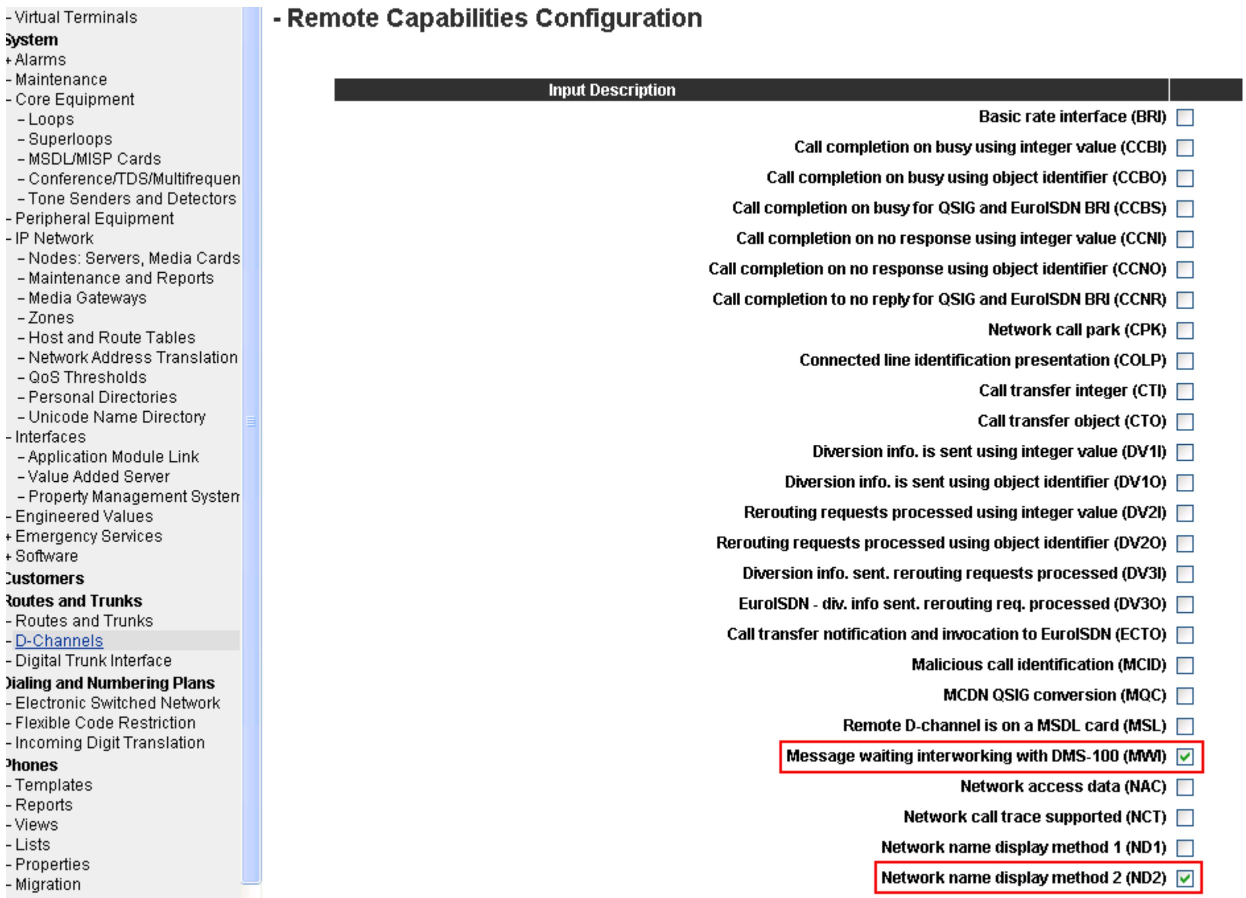


Figure 5 – D-Channel Basic Options

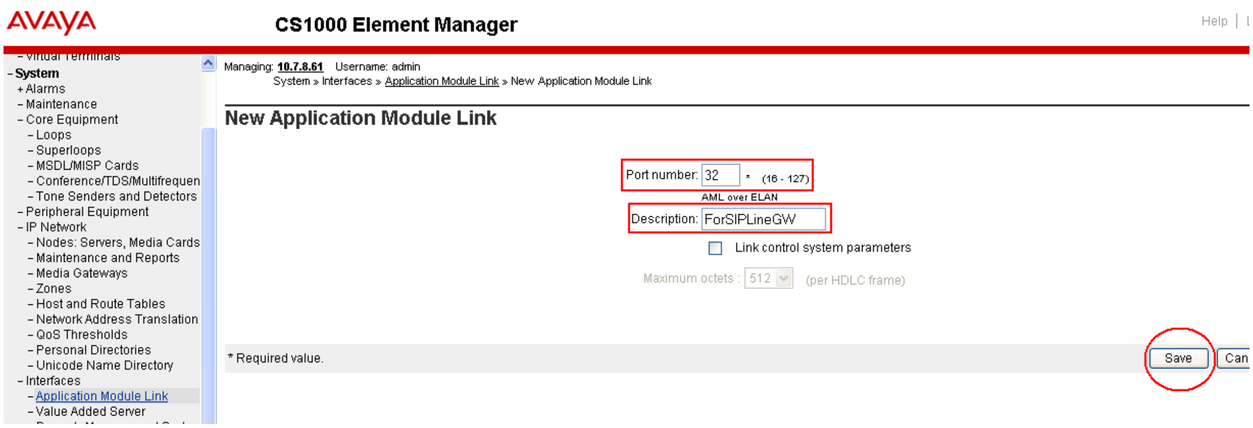
The **Remote Capabilities Configuration** page is displayed. Select the **Message waiting interworking with DMS-100 (MWI*)*** check box,[2] and the **Network name display method 2 (ND2)** check box. At the bottom of the **Remote Capabilities Configuration** page, click **Return - Remote Capabilities** (not shown), and the **D-Channel Property Configuration** page reappears. Click on **Submit** (see lower left in **Figure 5**).



[2] Note that although the Avaya B179 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.

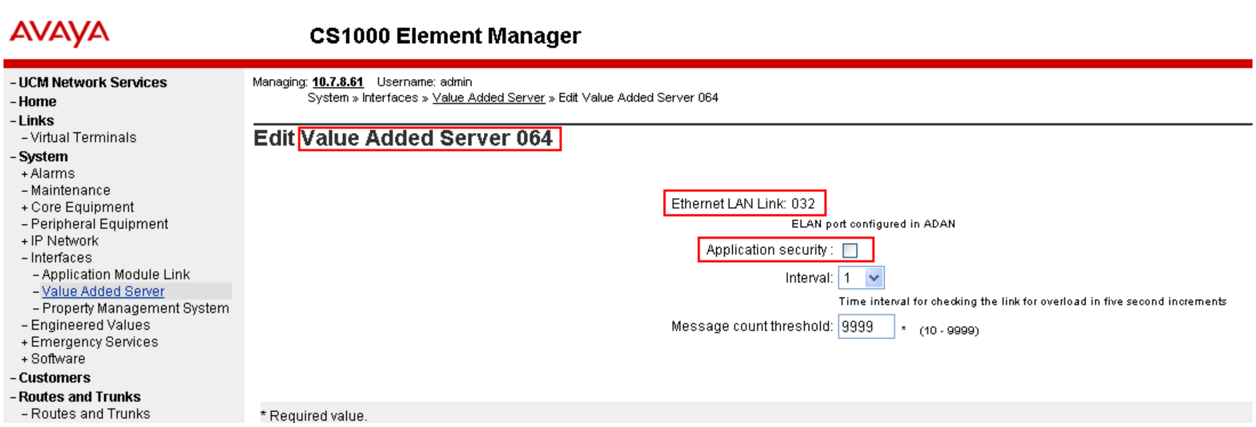
3.5. Configure Application Module Link (AML)

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.



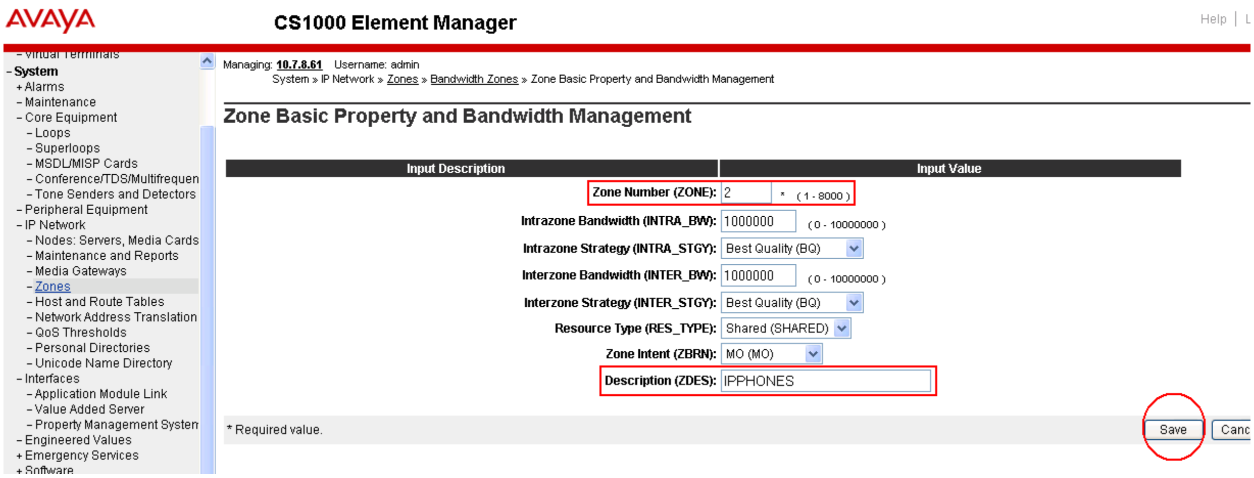
3.6. Configure Value Added Server (VAS)

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 (not shown), enter a **Value added server ID** (64 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.



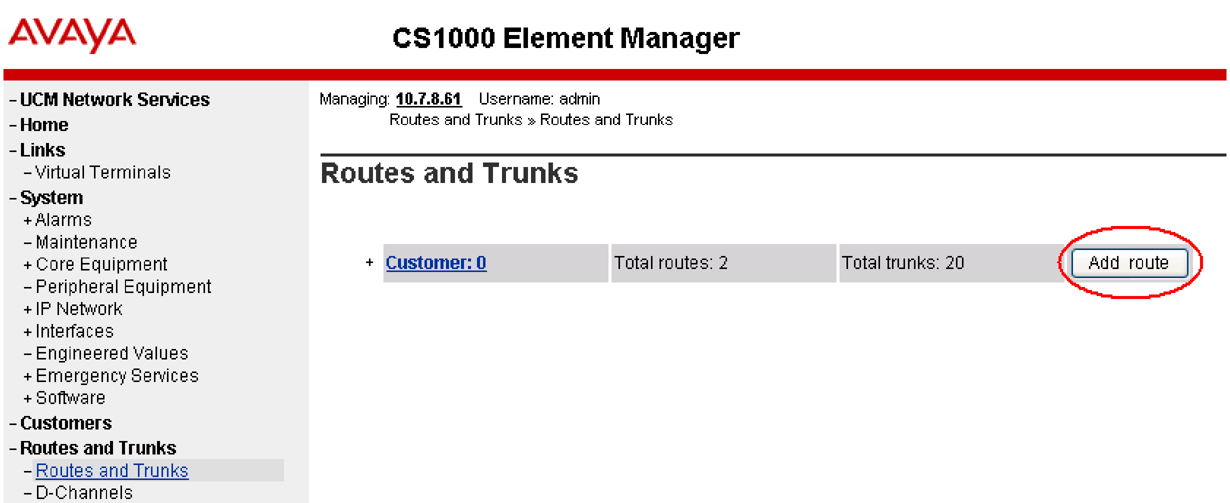
3.7. Configure Zone for SIP Phones

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), and on the **Zone Basic Property and Bandwidth Management** page, enter a **Zone number (ZONE)** and an appropriate **Description**. Defaults can be used for the remaining fields. Click **Save**.



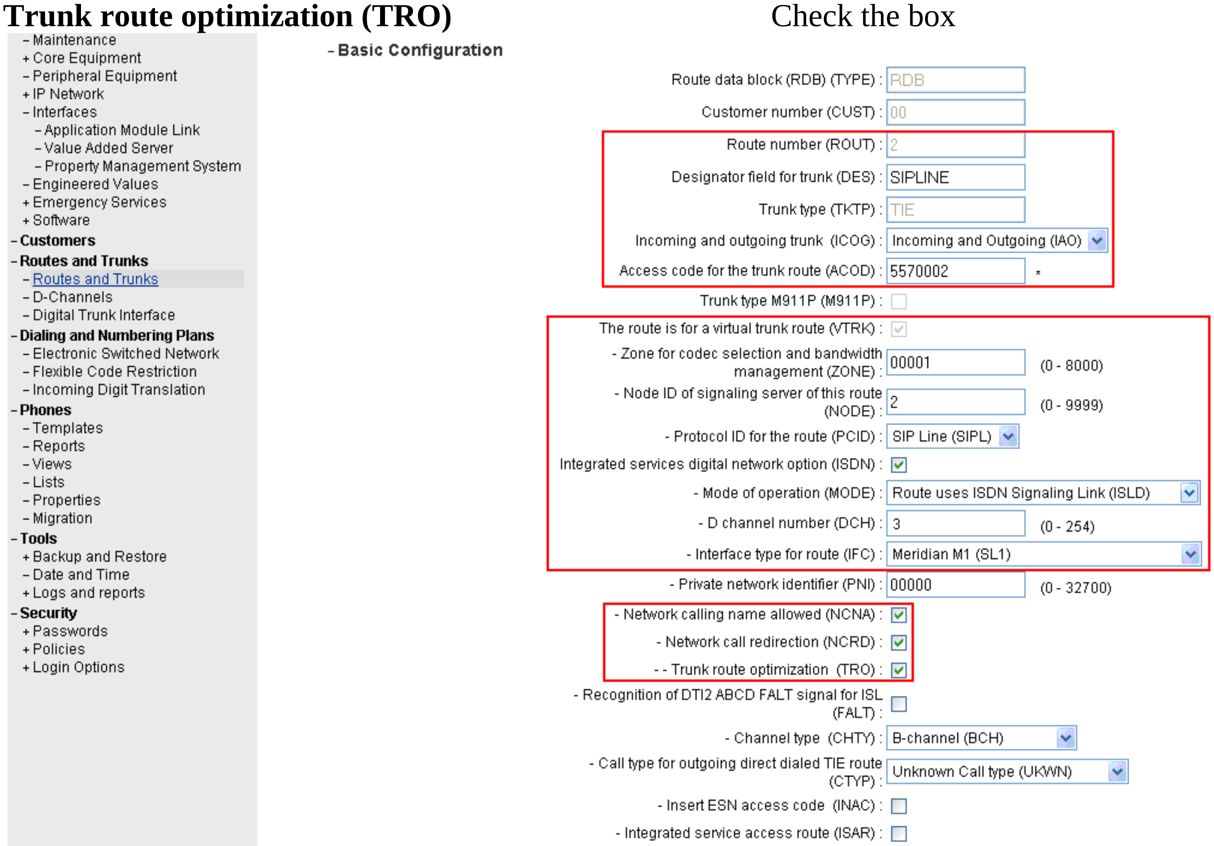
3.8. Configure SIP Line Route Data Block (RDB)

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 a 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[3]
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
Network call redirection (NCRD)	Check the box
Trunk route optimization (TRO)	Check the box

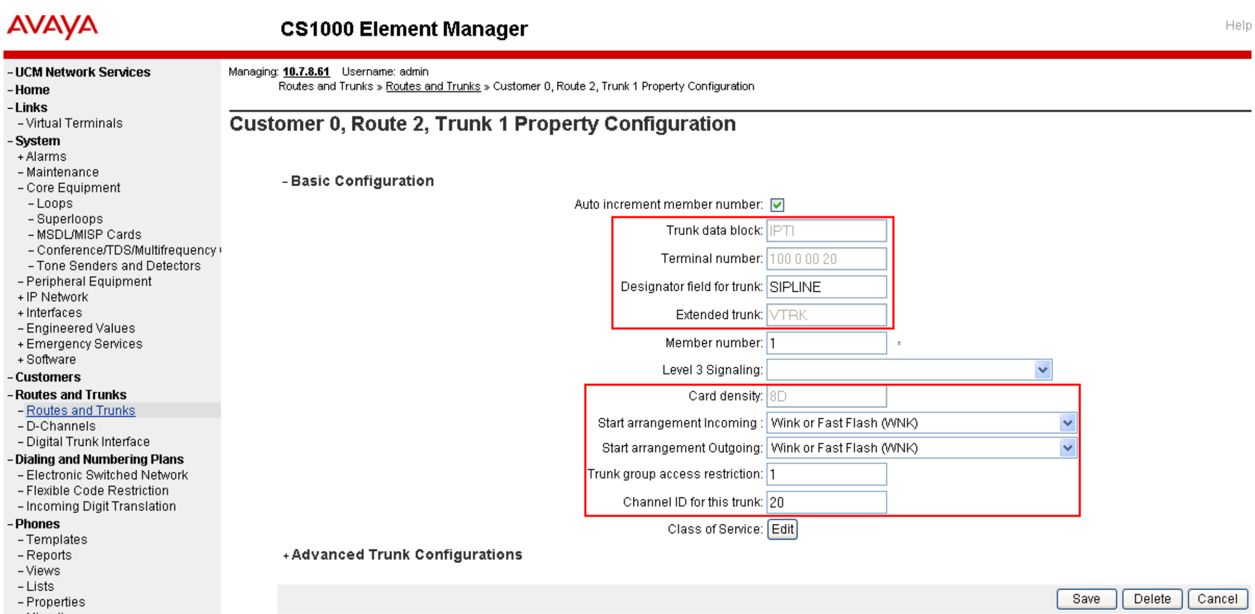


[3] Note that this must be a zone of type VTRK and must be different than the zone created for the SIP phones in **Section 3.7**. In the sample configuration, the VTRK zone was 1.

3.9. Configure SIP Line Virtual Trunk

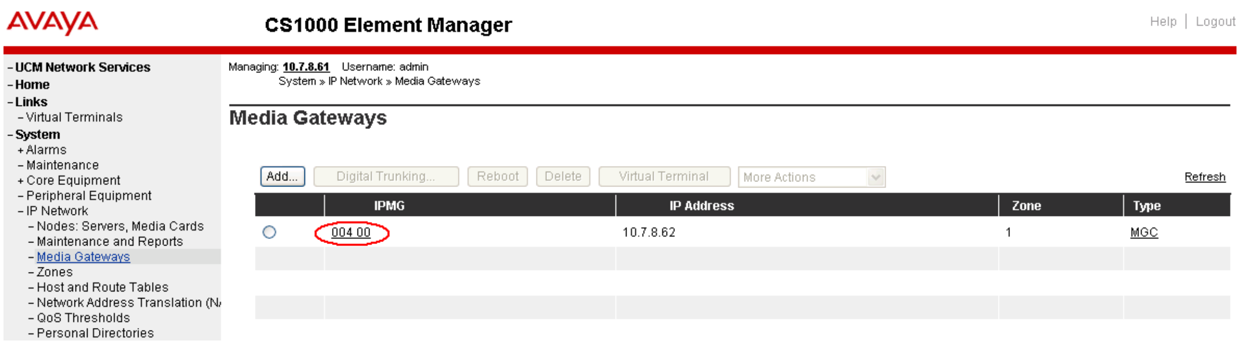
When the **Routes and Trunks** screen is displayed after adding the route in **Section 3.8**, click **Add trunk** corresponding to the newly added route to add new trunk members. The following screen shows the parameter settings for one of the trunks after they have been added. Set the following parameters 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 **Wink or Fast Flash (WNK)**
Start arrangement Outgoing	Select **Wink or Fast Flash (WNK)**
Trunk group access restriction	Desired trunk group access restriction level.
Channel ID for this trunk	An available starting channel ID.

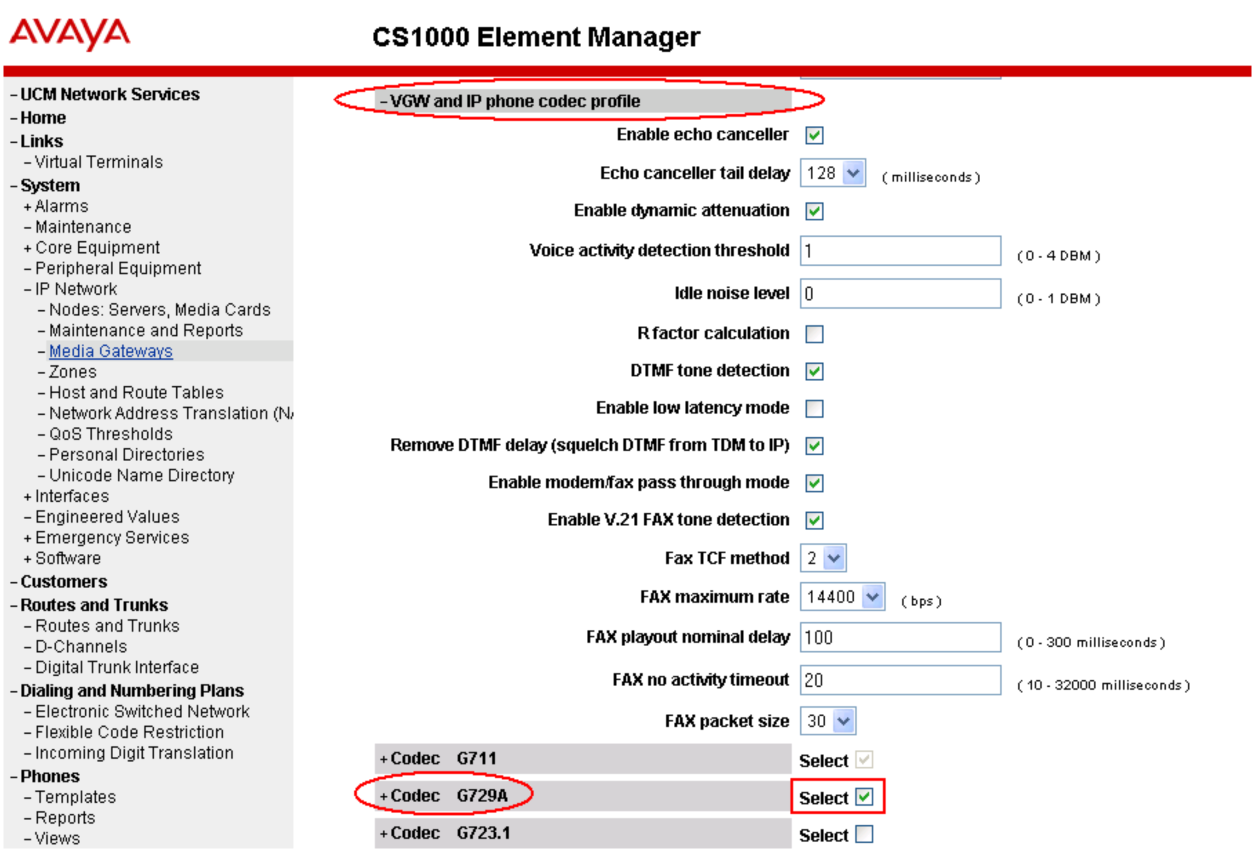


3.10. Configure Media Gateway Controller

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 (see **Section 4.2 Figure 9**). Click **Save**.

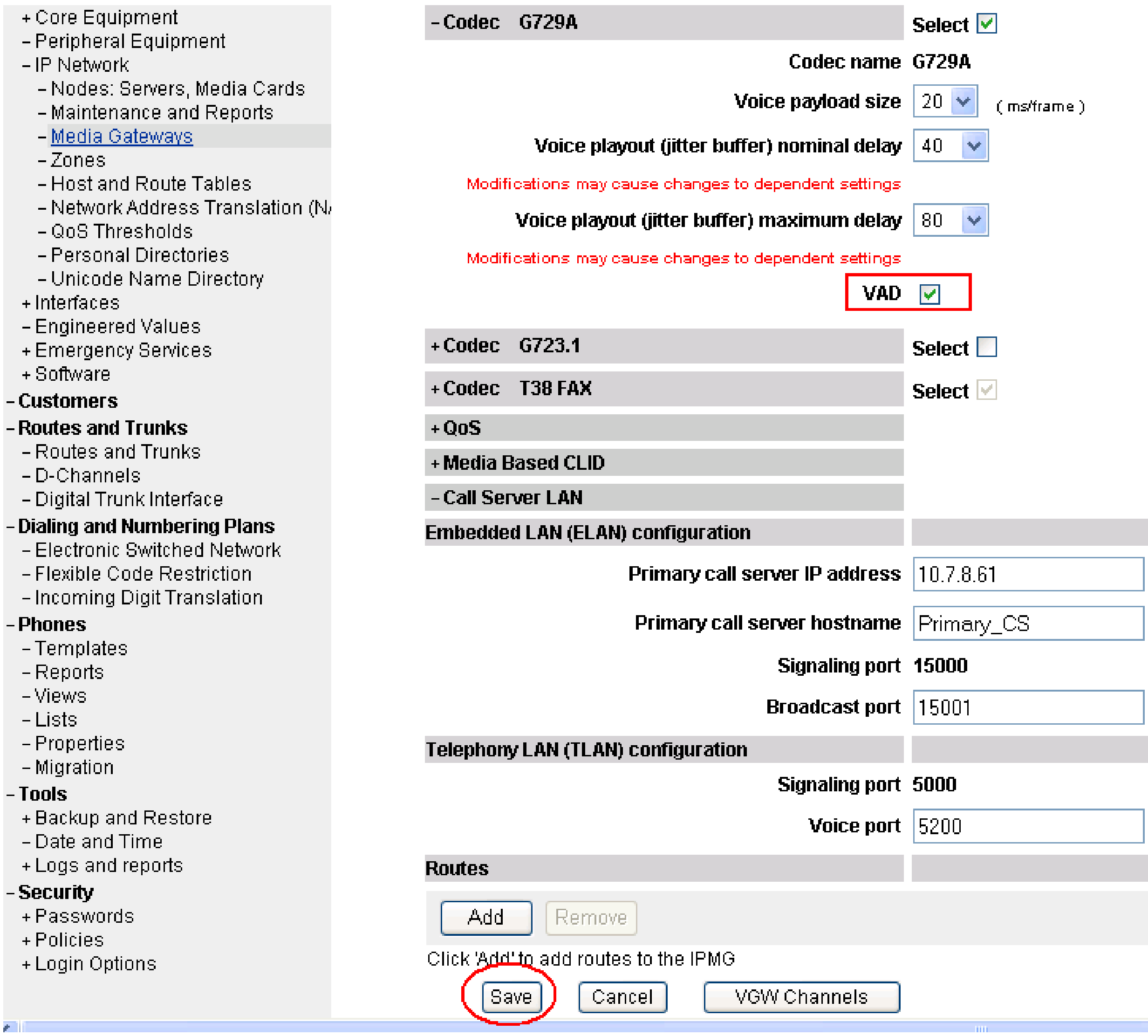
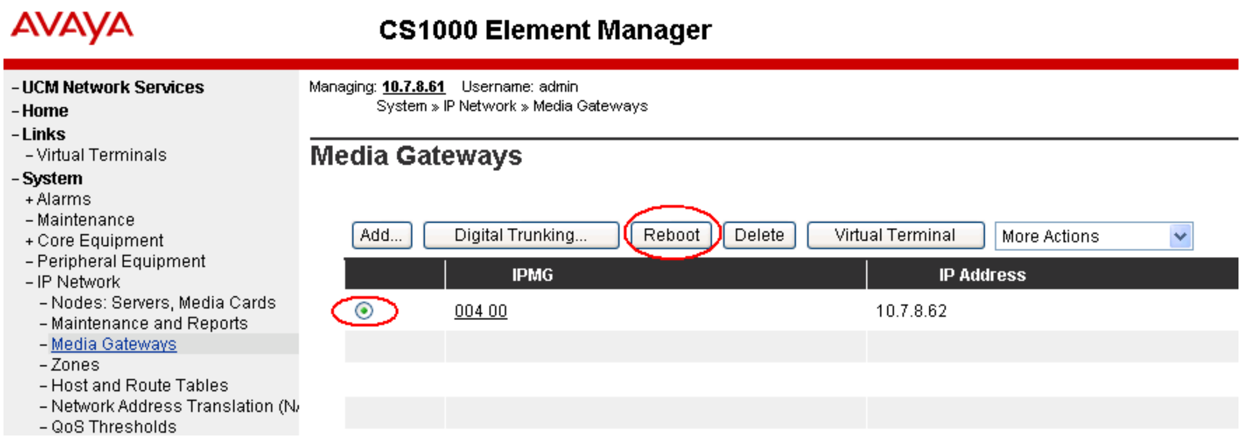


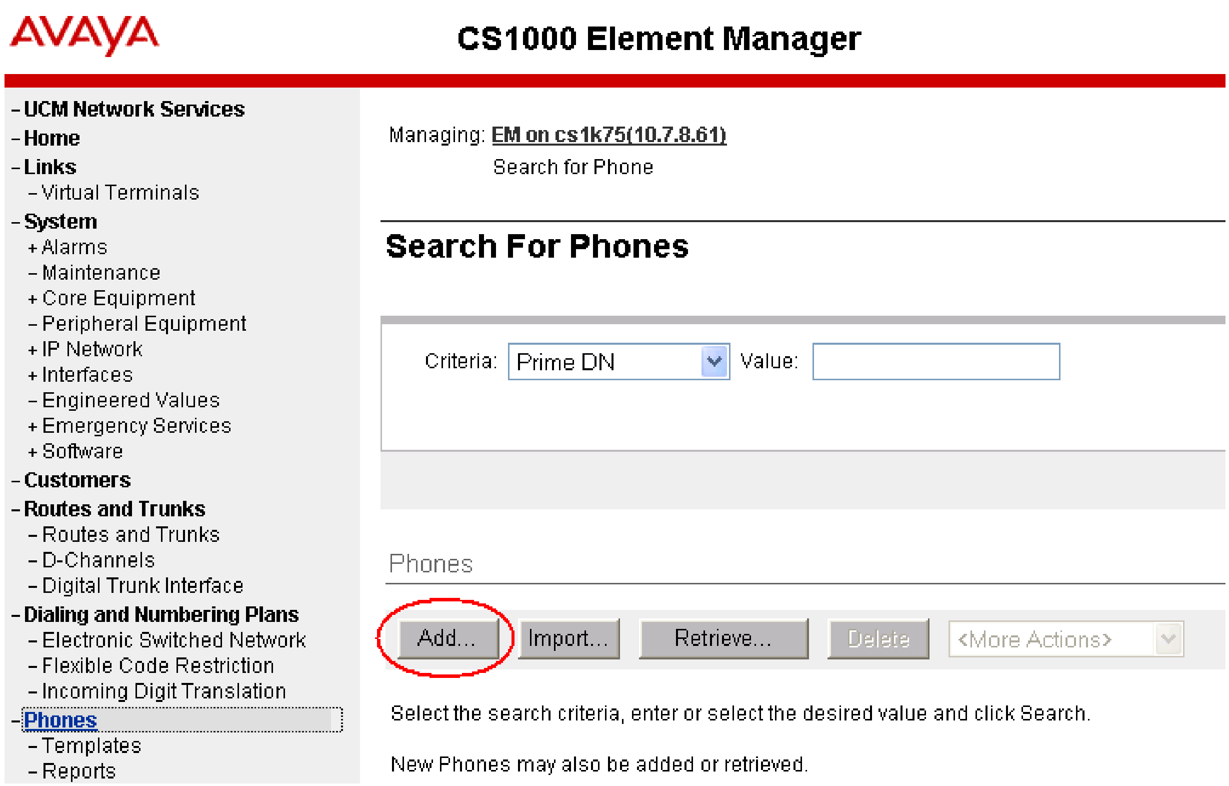
Figure 6 – MGC Codec Selection

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

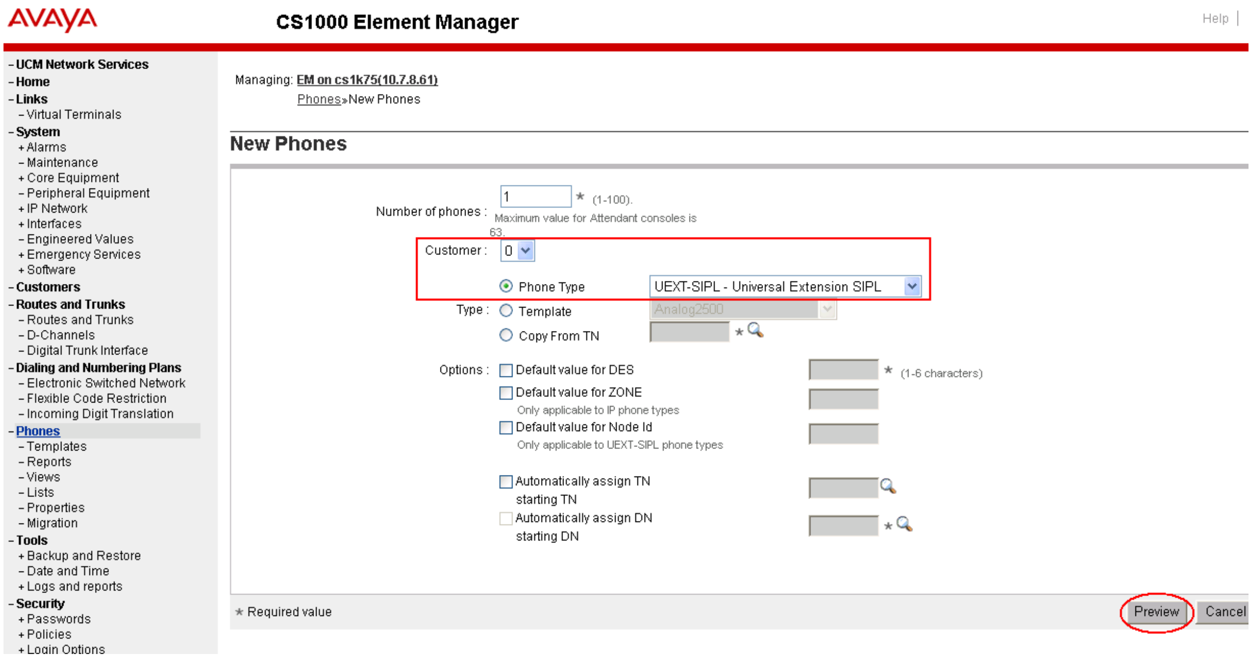


3.11. Configure SIP Line Telephone

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

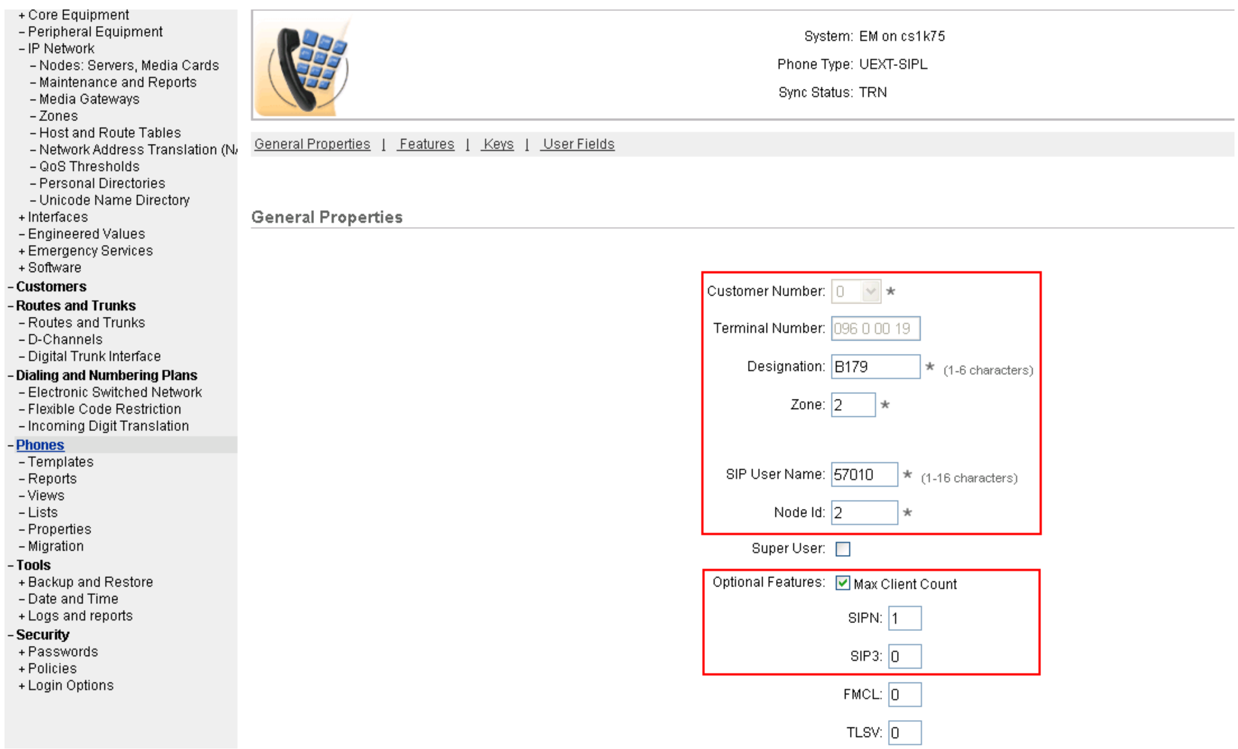


On the **New Phones** page, select the **Customer**, select the **Phone Type** radio button, and then select **UEXT-SIPL – Universal Extension SIPL**. Click **Preview**.



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 **1**
SIP3	Set to **0**

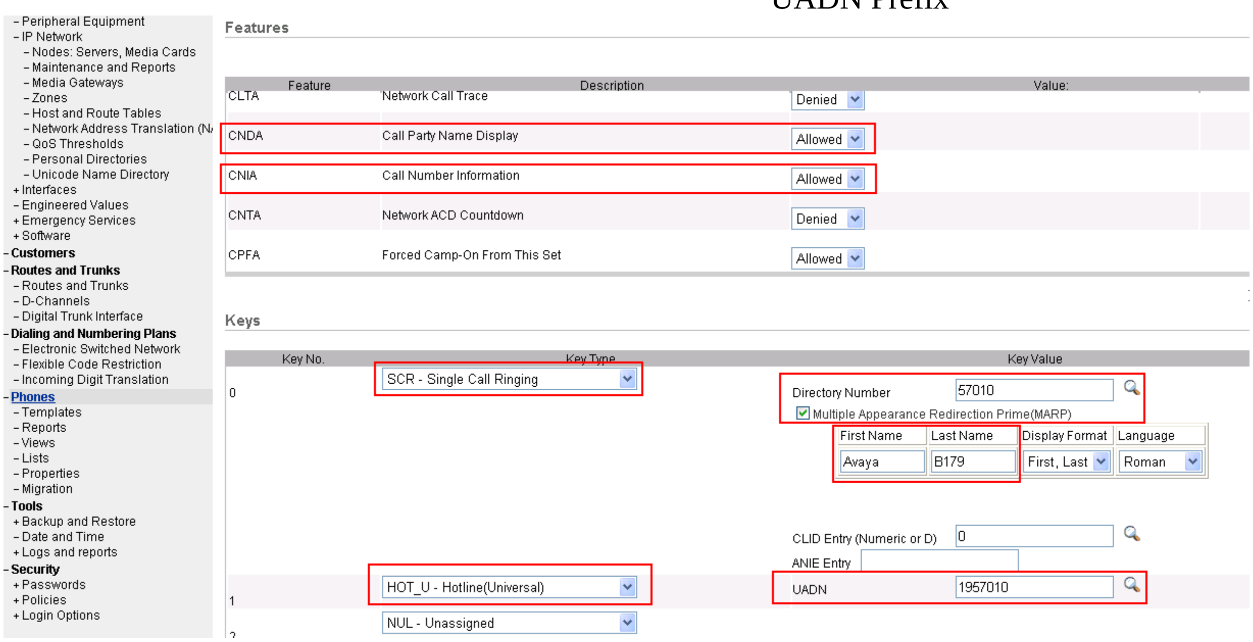


In the **Features** section, fill in the following fields, leaving the remaining fields at their defaults. Note that only the first two feature settings are shown in the screen below; the scroll bar must be used to display and set the remaining features, which are not shown here.

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[4]



[4] 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**.

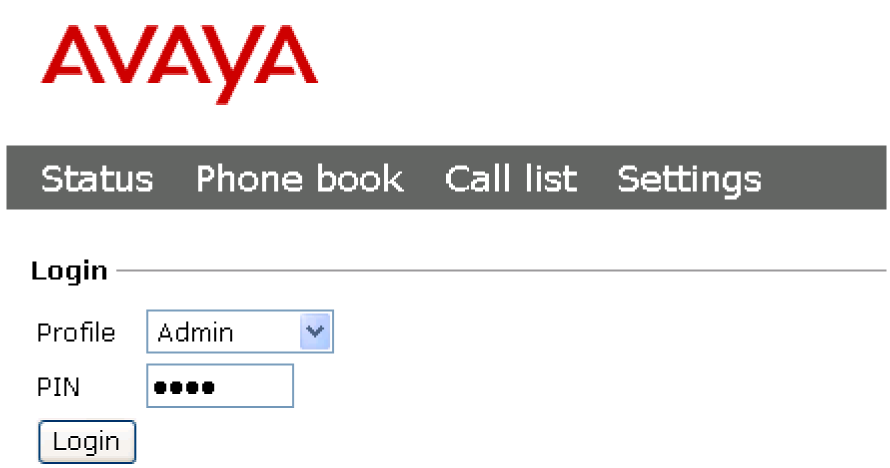
Click **Save** (not shown) to save the configuration for this phone.

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. Select **Admin** in the **Profile** dropdown list and enter the appropriate password.



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

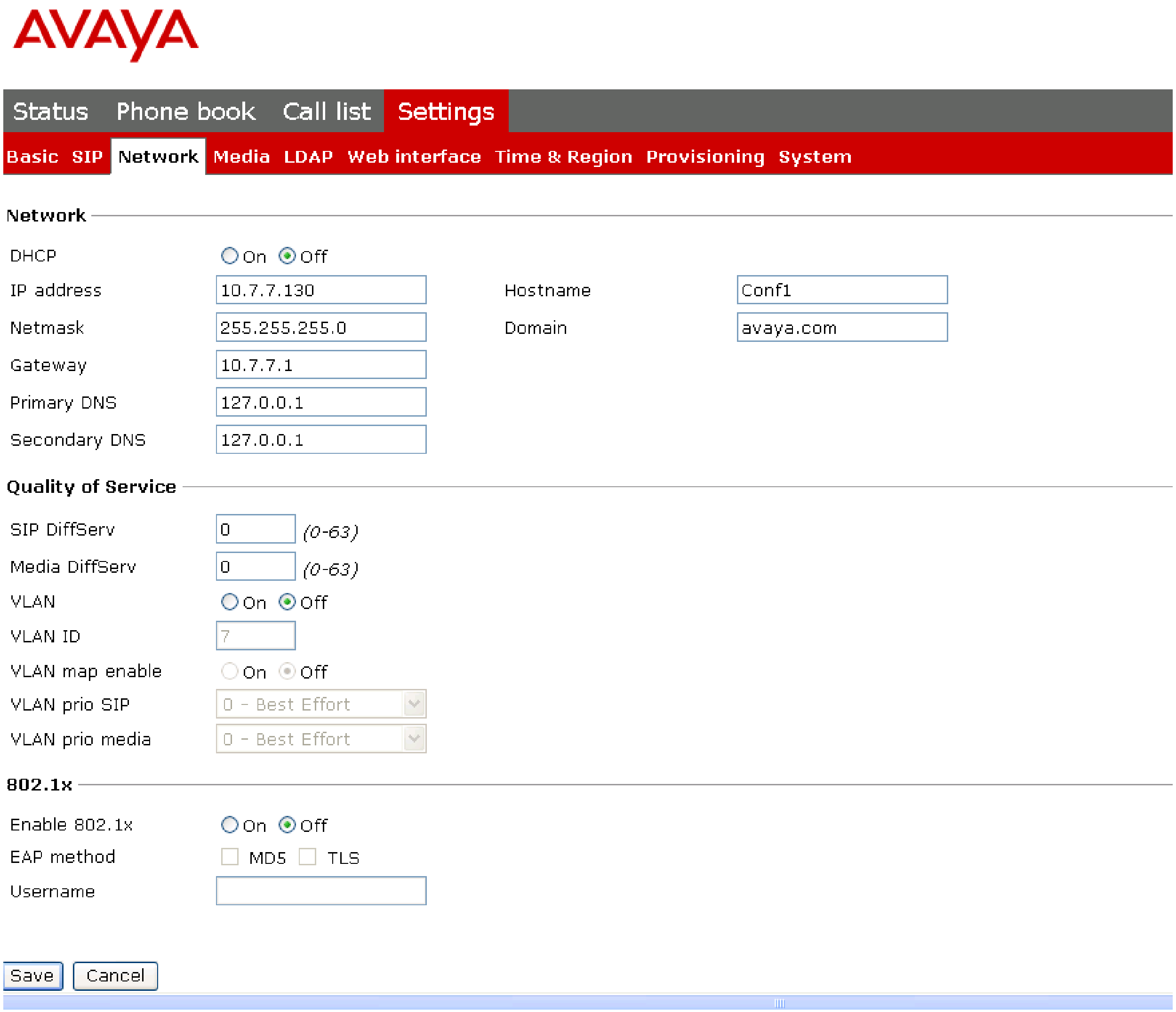


Figure 7 – B179 Network Configuration Status

4.2. Configure SIP Signalling Settings

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

Under **Account 1**:

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 "*"
Authentication name	Extension (**SIP User Name**) of the SIP Line telephone configured in **Section 3.11**
Registrar and **Proxy**	SIP domain configured in the CS 1000
Password	The **Station Control Password** of the SIP Line telephone configured in **Section 3.11**
Registration interval	Enter a value (1800 was used in the sample configuration)

Under **Advanced**:

Enable Blind Transfer	Select the **No** radio button[5]
Outbound proxy	Enter the IP address of the CS 1000 Node (see **Figure 2**), port **5070**, and **lr** (loose routing), as shown

Under **Transport**:

Protocol	Select the **TCP** or **UDP** radio button (UDP shown)
Local UDP Port	Enter **5060**

Click **Save**.

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

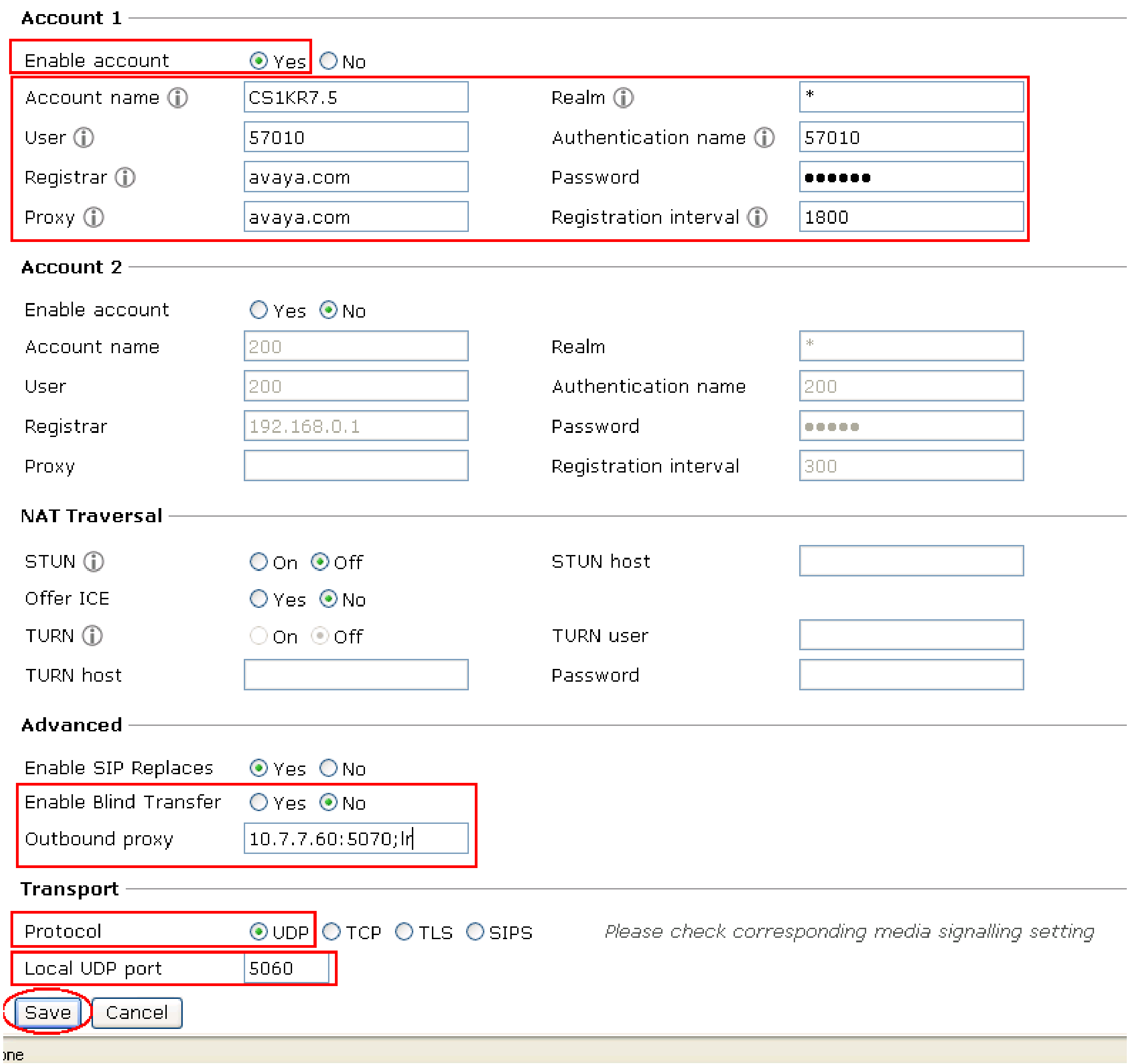
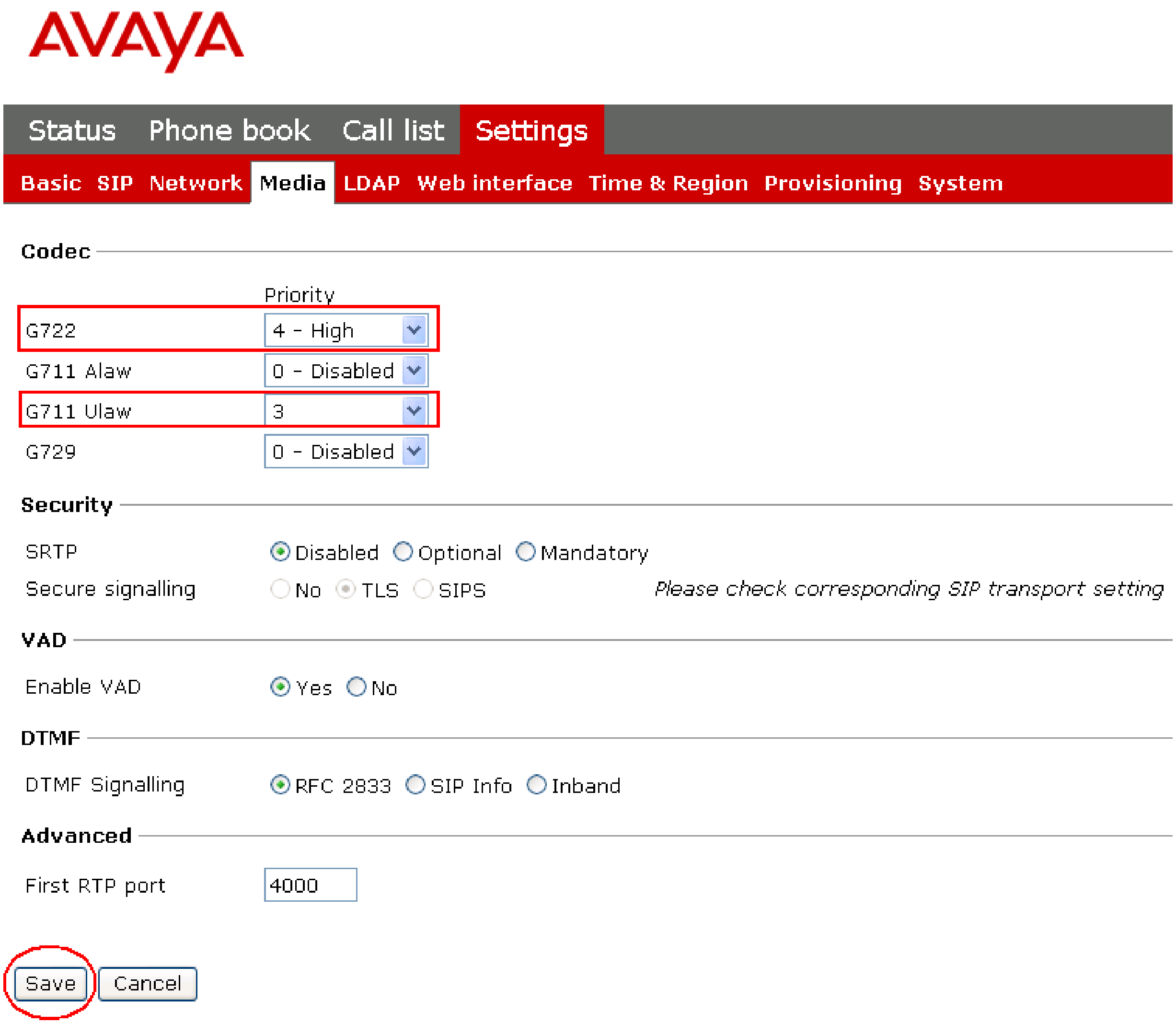


Figure 8 – B179 SIP Configuration

To configure the audio codec settings, navigate to **Settings → Media**, and select the priority for codec selection. One combination shown below prefers the high fidelity G.722 codec if the other party's telephone can support it, with a fallback to G.711. Defaults can be used for the remaining fields. Note that call transfer by CS 1000 telephones is not supported for G.722.



Another combination that could be used in low bandwidth environments would be to give high preference to the compressed G.729A codec, with fallback to G.711, as shown below. In the case of G.729A, ensure that the **VAD** setting matches that configured in the CS 1000. In this case, checking **Enable VAD** results in G.729AB. After choosing an appropriate codec preference list, click **Save**. Note also that for this release of the Avaya B179, G.711 is required to be in the codec list with G.729 in order for call hold by CS 1000 telephones to operate correctly.

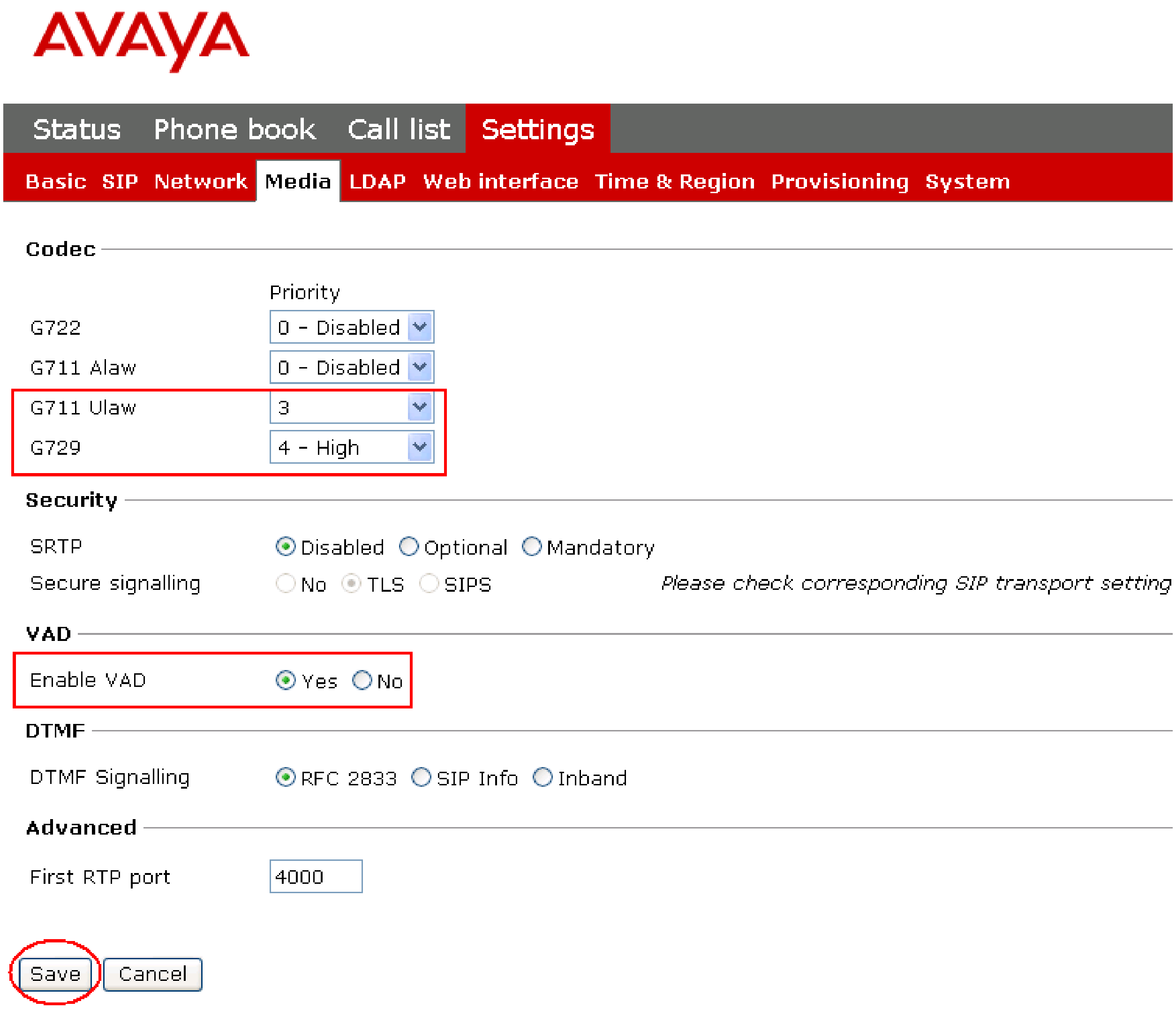


Figure 9 – B179 Codec Selection

After the configuration has been saved, the B179 will register with the CS 1000, and a display similar to those shown in **Figures 10** and **11** below will appear on the telephone. The **Hostname** (see **Figure 7**) is displayed at the center, and in the lower left corner is the **Account name** (see **Figure 8**). 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 (**Figure 10**), the B179 has successfully registered. If the square is not filled in (**Figure 11**), registration was unsuccessful.

Figure 10 – Successful B179 Registration

Figure 11 – Unsuccessful B179 Registration

5. Verification Scenarios

Verification scenarios for the configuration described in these Application Notes included:

- Registration and recovery, including power cycling and network disruption.
- SIP signaling using UDP and TCP transport
- Basic calling among the B179 and the following CS 1000 supported telephones:
    - 1120e UNIStim
    - 1165e SIP
    - 2007 UNIStim
    - 2050PC UNIStim (soft phone)
    - M3903 Digital
- RFC 2833 DTMF support
- G.711mu-law, G.722, G.729A, and G.729AB audio codec support.
- Hold, consultative hold.
- Manual conference by the B179.
- Unattended transfer.
- Placement of calls via the outbound call log.

The following restrictions to the above features apply:

- Attended call transfer of the B179 by CS 1000 supported telephones is supported, except for M3900 series digital telephones and when G.722 codec is used.. Attended call transfer by the B179 is not supported.
- Call hold by CS 1000 supported telephones is supported for G.729 only if G.711 is included in the B179 codec list.
- Calls from the B179 via the inbound call log are not supported.
- Group conference by the B179 is not supported.

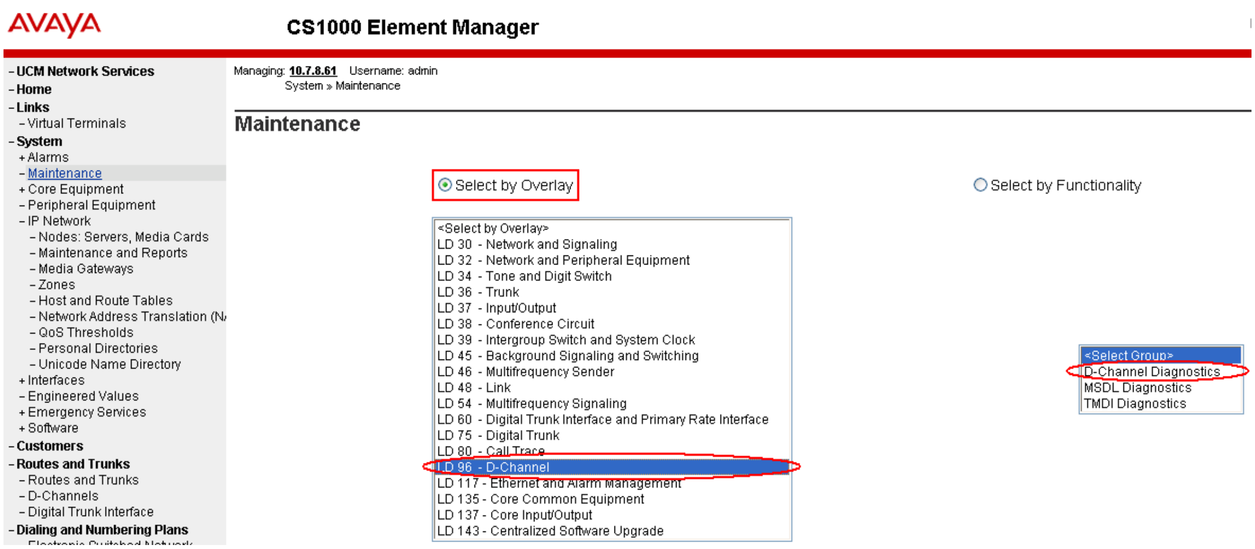
6. Verification Steps

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

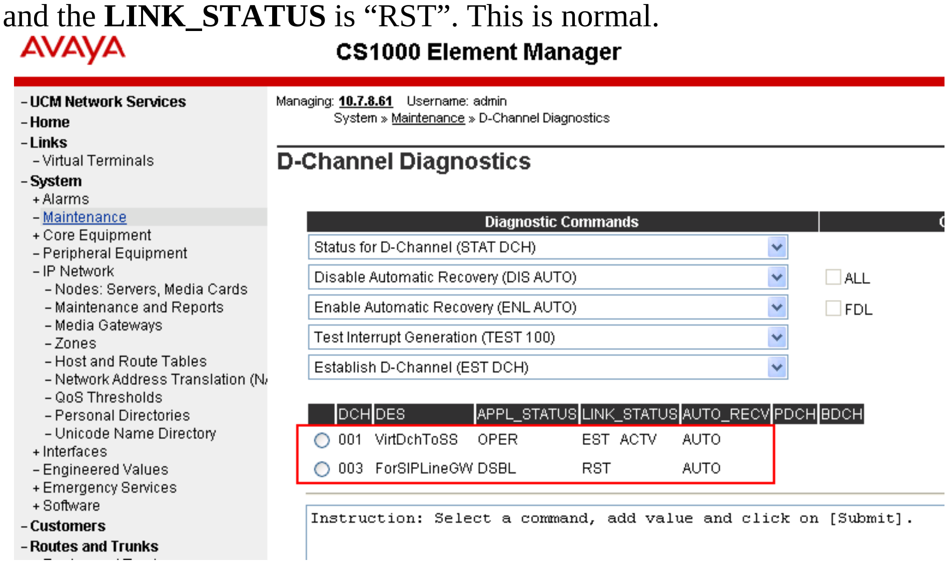
6.1. Verify Avaya Communication Server 1000

6.1.1. Verify D-Channel Status

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

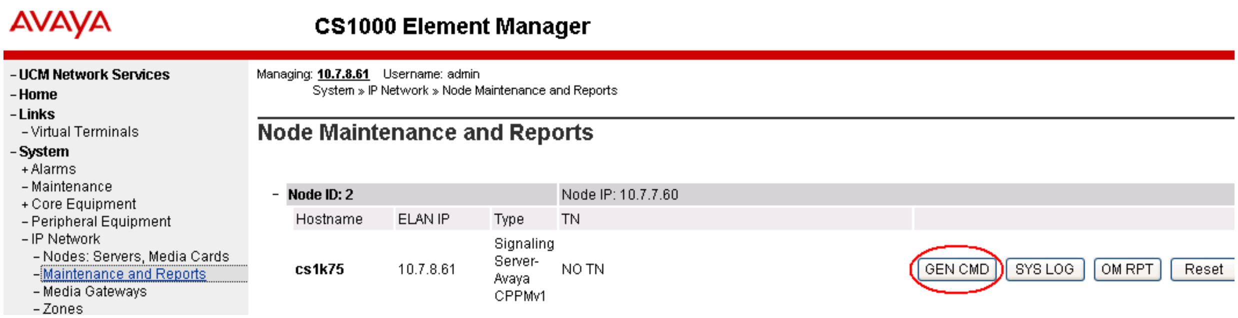


The screen below shows the **APPL_STATUS** of the SIP trunk D-Channel as "OPER" and the **LINK_STATUS** as "EST ACTV". Note that for the SIP line D-Channel, the **APPL_STATUS** is "DSBL" and the **LINK_STATUS** is "RST". This is normal.

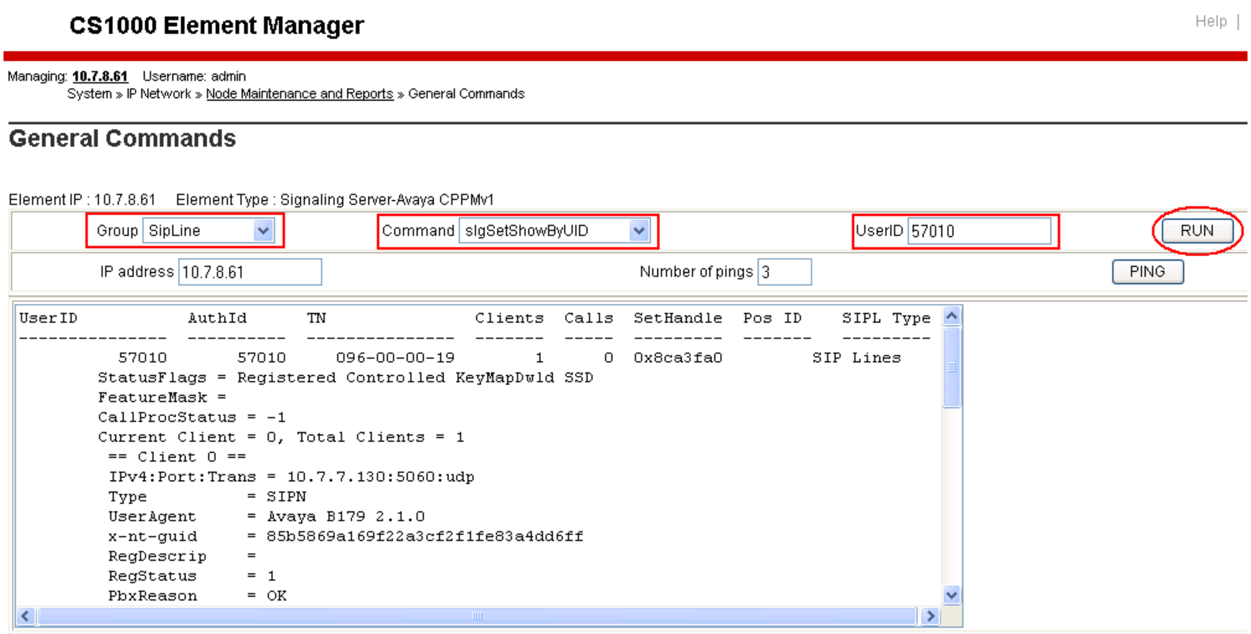


6.1.2. Verify SIP Registration Status

In the Element Manager web interface, navigate to **System → IP Network → Maintenance and Reports** on the left pane. Click **GEN CMD**.

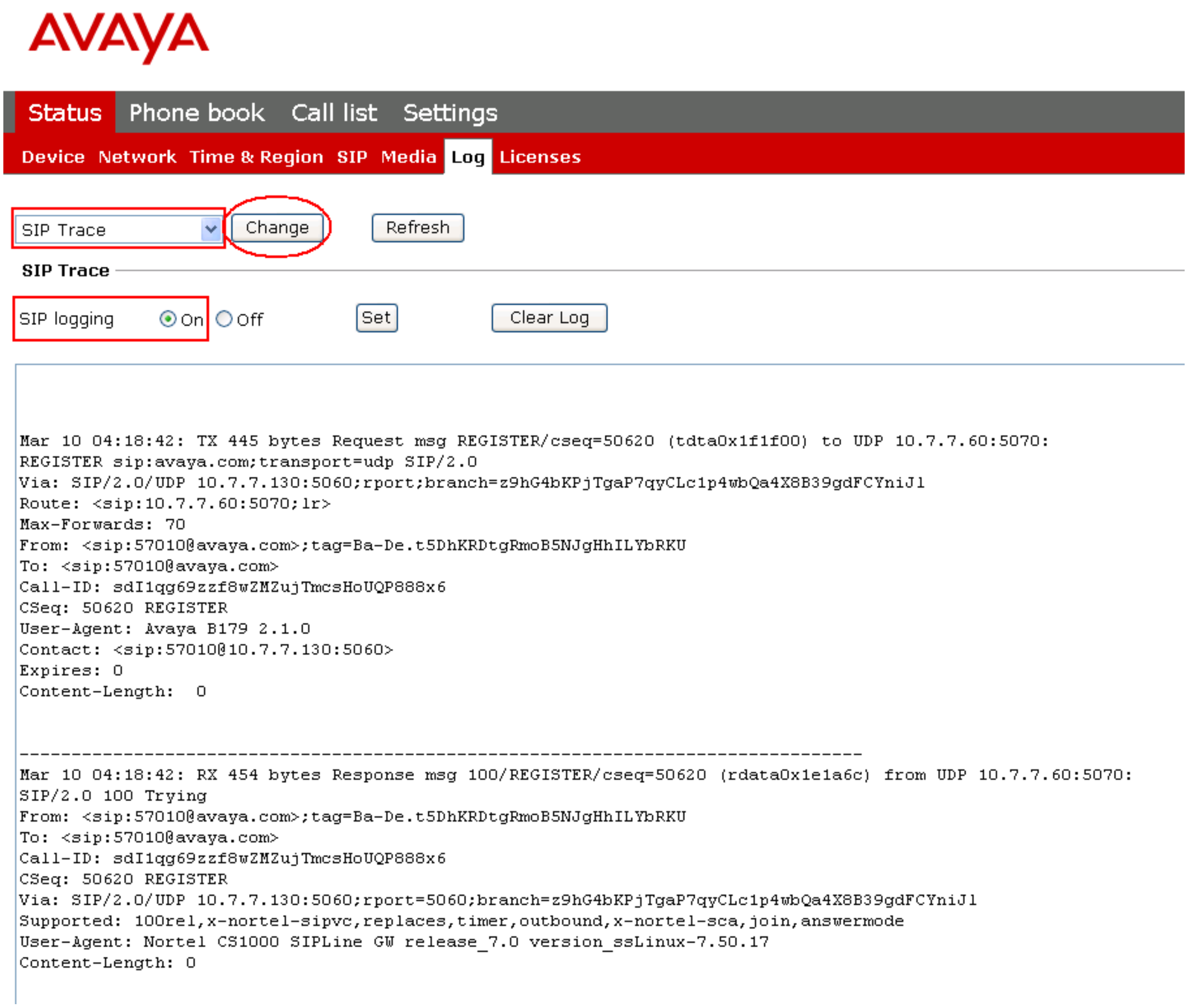


The **General Commands** page is displayed. From the **Group** drop-down menu select **SipLine**, from the **Command** drop-down menu select **slgSetShowByUID**, enter the B179 extension in **UserID**, and click on **RUN**. The output shown indicates successful registration and displays details of the registration parameters. Note that if the B179 has not registered, the error message "Invalid userId 57010" will be returned instead of the detailed registration information.



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. See **Figures 10** and **11** for examples of successful and unsuccessful registration. 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 **On** radio button of the **SIP logging** field 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 and the Avaya B179 SIP Conference Phone can be used together in an integrated solution supporting the features described in **Section 5**.

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: 7.5, Document Revision: 05.04, Document #NN43001-632.

[2] *Communication Server 1000 SIP Line Fundamentals*, Release 7.0, Document #NN43001-508, 02.03, August 2010.

[3] *Installation and Administration of B179*, 110047-61-001, Rev 3d.

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