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Avaya Solution & Interoperability Test Lab

Application Notes for Algo 8028 SIP Doorphone with Avaya Communication Server 1000 SIP Line Release 7.5 – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Server 1000 SIP Line Release 7.5 and Algo 8028 SIP Doorphone. During the compliance testing, the Algo 8028 controller was able to register as a SIP client endpoint with the Communication Server 1000 SIP Line gateway. The Algo 8028 SIP Doorphone was able to place and receive calls from the Communication Server 1000 Release 7.5 non-SIP and SIP Line telephones.

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 detailed configurations for the Avaya Communication Server 1000 SIP Line release 7.5 (hereafter referred to as CS1000) and the Algo 8028 SIP Doorphone (hereafter referred to as Algo 8028) firmware version 1.1.4 used during the compliance testing. All the applicable telephony feature test cases of release 7.5 SIP line were executed on the Algo 8028 , where applicable, to ensure interoperability with the CS1000.

2. General Test Approach and Test Results

The general test approach was to have the Algo 8028 register to the CS1000 SIP line gateway successfully. From the CS1000 telephone clients/users calls were placed to and from the Algo 8028 and other telephony features such as busy, hold, DTMF, MWI and codec negotiation were tested.

2.1. Interoperability Compliance Testing

The focus of this testing was to verify that the Algo 8028 SIP Doorphone was able to interoperate with the CS1000 SIP line system. The following areas were covered:

- Registration of the Algo 8028 to the CS1000 SIP Line Gateway.
- Call establishment of Algo 8028 with CS1000 SIP and non-SIP telephones, and ensure two way speech paths.
- CS1000 telephones can send DTMF tones to Algo 8028 controller to exercise the remote door release function.
- Telephony features: Basic calls, conference, blind and consultative transfer (from Avaya phone), DTMF (dual tone multi frequency) via RFC2833, busy, mute, call forward no answer, busy and all calls.
- PSTN calls over a PRI trunk.

2.2. Test Results

The objectives outlined in the **Section 2.1** were verified. The following observations were made during the compliance testing:

- Avaya has not performed audio performance testing or reviewed the Algo 8028 compliance to required industry standards.
- It is highly recommended to disable the media security on the Call Server to avoid some unexpected behaviors such as one way audio from a call made from the PSTN over a PRI trunk.

2.3. Support

For technical support on Algo 8028, please contact Algo technical support team:

- **Telephone:** 1-877-884-2546
- **Email:** support@algosolutions.com
- **Website:** www.algosolutions.com/8028

3. Reference Configuration

Figure 1 illustrates the test configuration used during the compliance testing between the Avaya CS1000 and the Algo 8028.
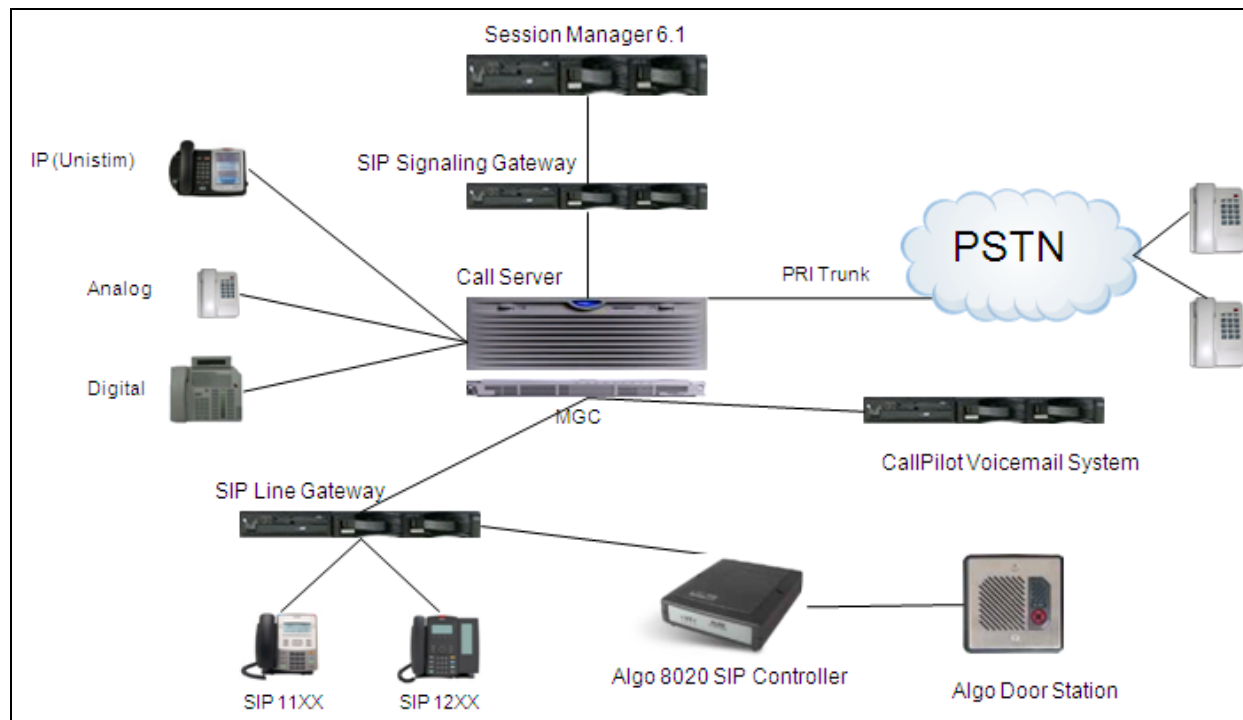

Figure 1: Network Configuration Diagram

4. Equipment and Software Validated

The following equipment and software was used during the lab testing:

Equipment	Software Version
Avaya CS1000E	Call Server (CPPM): 7.50Q Signaling Server (CPPM): 7.50.17
Avaya CallPilot™ Messaging System	5.0.1
Avaya IP Soft Phone 2050 Avaya IP Phone 1140 Avaya IP Phone 2004P2 Avaya IP Phone 2002P2	3.04.0003 0625C6O 0692D93 0604DC5
Avaya SIP 1140	02.02.21.00
Algo 8028 SIP Door Phone	Firmware version 1.1.4

5. Configure Avaya CS 1000 - SIP LINE

This section describes the steps to configure the Avaya CS1000 SIP Line using the CS1000 Element Manager. A command line interface (CLI) option is available to provision the SIP Line application on the CS1000 system. For detailed information on how to configure and administer the CS1000 SIP Line, please refer to **Section 9 [1]**.

The following is a summary of tasks that need to be done to configure the CS1000 SIP Line:
- Log in to Unified Communications Management (UCM) and Element Manager (EM).
- Enable SIP Line Service and Configure the Root Domain.
- Create SIP Line Telephony Node.
- Create D-Channel for SIP Line.
- Create an Application Module Link (AML).
- Create a Value Added Server (VAS).
- Create a Virtual Trunk Zone.
- Create a Route Data Block (RDB).
- Create SIP Line Virtual Trunks.
- Create SIP Line phones.

5.1. Prerequisite

This document assumes that the CS1000 SIP Line server has:
- Been installed with CS1000 Release 7.5 Linux Base.
- Joined CS1000 Release 7.5 Security Domain.
- Been deployed with SIP Line Application.

The following packages need to be enabled in the key code. If any of these features have not been enabled, please contact your Avaya account team or Avaya technical support at http://www.avaya.com.

Package Mnemonic	Package #	Descriptions	Package Type	Applicable market
SIP_LINES	417	SIP Line Service package	New package	Global
FFC	139	Flexible Feature Codes	Existing package	Global
SIPL_AVAYA	415	Avaya SIP Line package	Existing package	Global
SIPL_3RDPARTY	416	Third-Party SIP Line Package	Existing package	Global

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

Use the Microsoft Internet Explorer browser 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.

Log in with the username/password which was defined during the primary security server configuration, the UCM home page appears as shown in the **Figure 2** below.

Figure 2: The UCM Home Page of CS 1000 Release 7.5

On the UCM home page, under the **Element Name** column, click on the EM name of the CS1000 system that needs to be configured, in this sample that is **EM on cpppm3**. The CS1000 Element Manager page appears as shown in **Figure 3** below.
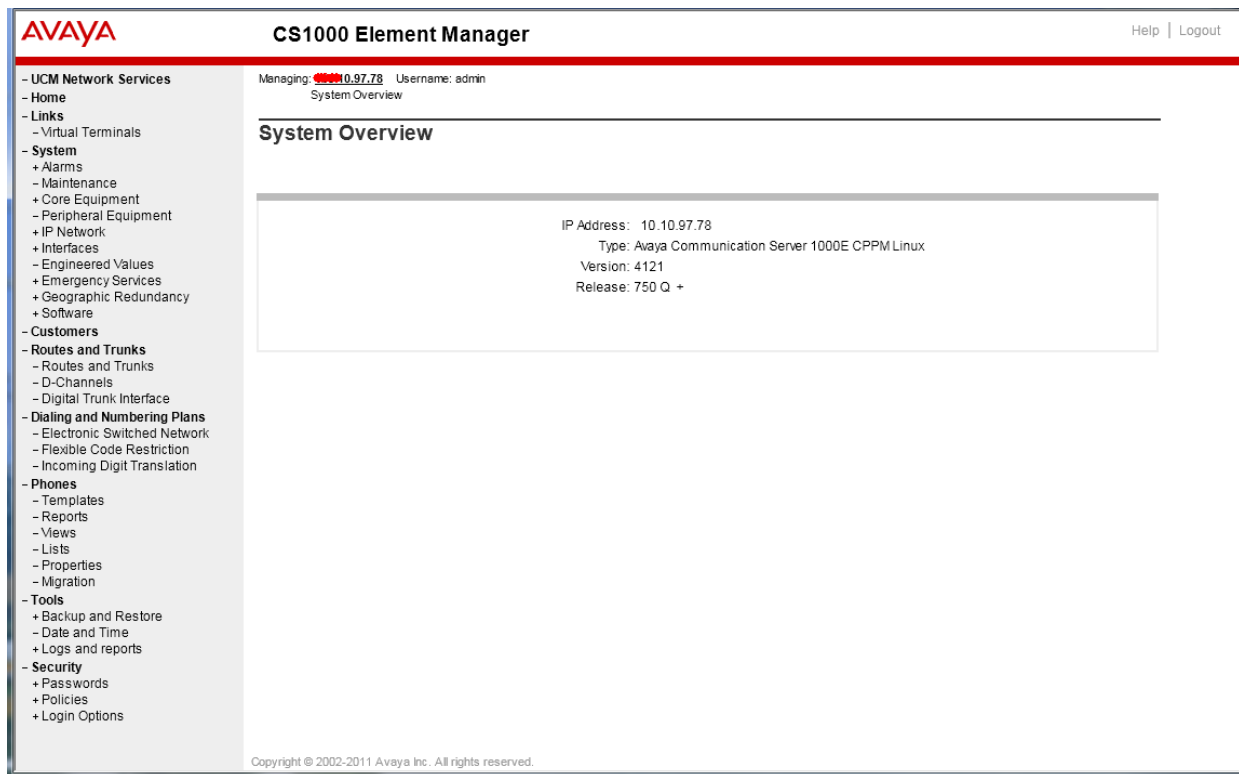
Figure 3: CS 1000 Release 7.5 EM Home Page

5.3. Enable SIP Line Service in the Customer Data Block

On the EM page, navigate to **Customers** on the left column menu (not shown); select the customer number to be enabled with SIP Line Service (not shown).
- Enable SIP Line Service by clicking on the **SIP Line Service** check box.
- Enter the prefix number in the **User agent DN prefix** text box as shown in **Figure 4**.
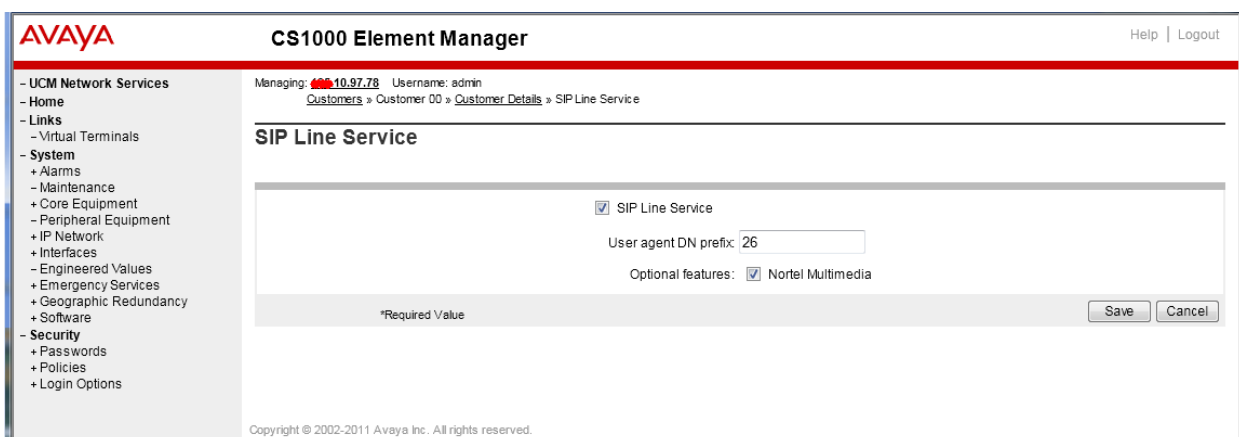
Figure 4: SIP Line Service in Customers Data Block

5.4. Add a new SIP Line Telephony Node

On the EM page, navigate to menu **System → IP Network → Nodes: Servers, Media Cards**. Click **Add** to add a new SIP Line Node to the IP Telephony Nodes (not shown). The new IP Telephony Node page appears as shown in **Figure 5**.

Enter the information as shown below:
- **Node ID** text box: 512 **->** this is the node ID of the SIP Line server.
- **Call Server IP Address** text box: 10.10.97.78.
- **Node IPv4 Address** text box: 10.10.97.187 **->** this is the IP address that a SIP endpoint uses to register to.
- **Subnet Mask** text box: 255.255.255.192.
- **Embedded LAN (ELAN) Gateway IP Address** text box: 10.10.97.65.
- **Embedded LAN (ELAN) Subnet Mask** text box: 255.255.255.192.
- Check **SIP Line** check box to enable SIP Line for this Node.

Figure 5: Adding a New IP Telephony Node

- Click on the **Next** button to go to next page**.** The page, New IP Telephony Node with Node ID, will appear as shown in **Figure 6**.
- On the **Select to Add** drop down menu list, select the desired server to add to the node.
- Click the **Add** button
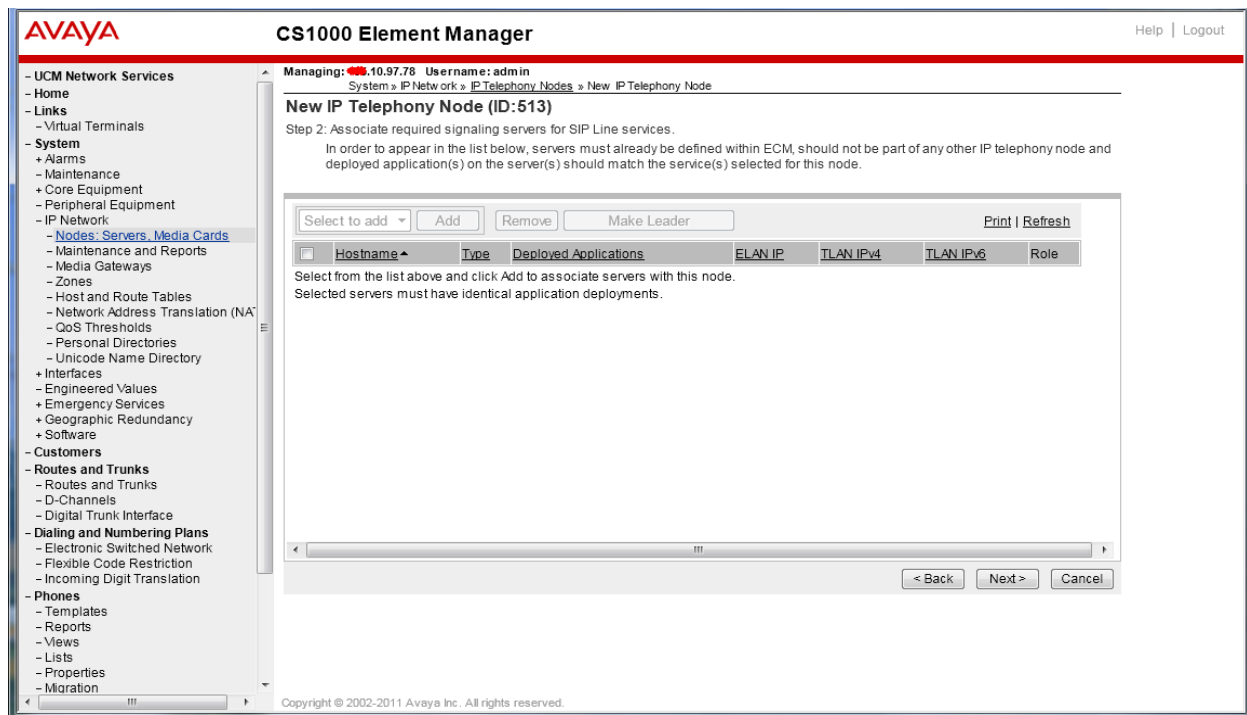- Select the check box next to the newly added server, and click **Make Leader** (not shown).

Figure 6: Adding a New IP Telephony Node (cont)

- Click on the **Next** button to go to the next page. The **SIP Line Configuration Detail** page appears as shown in **Figure 7**.
- Enter the SIP Line domain name in the **SIP Domain name** text box, for example **sipl75.com**.
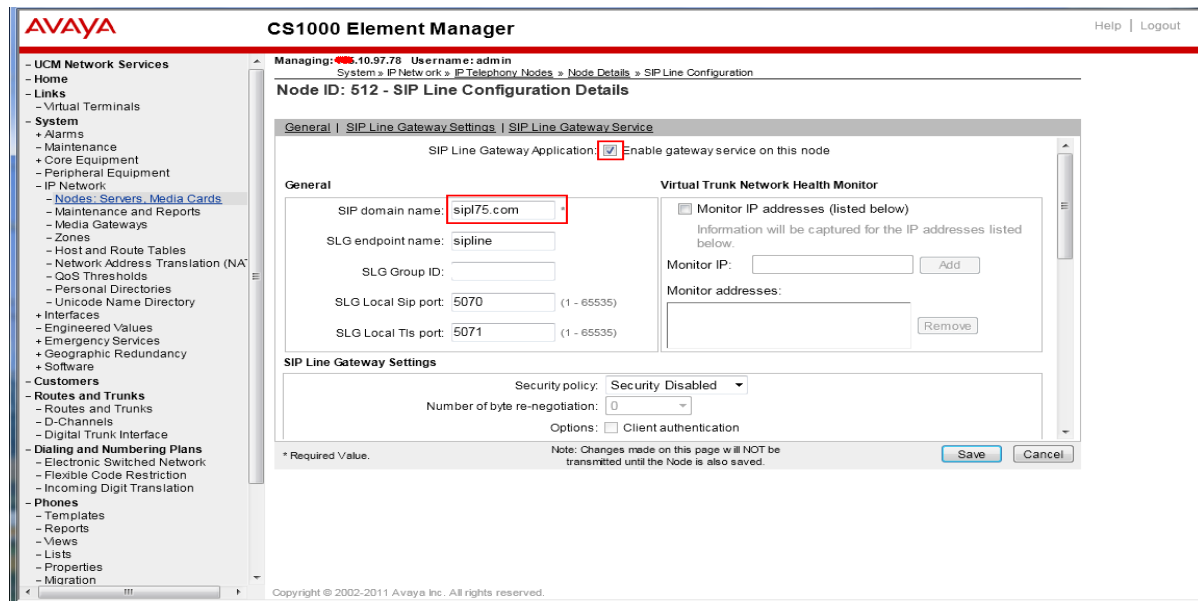

Figure 7: Adding a new IP Telephony Node (cont)

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- Under the **SIP Line Gateway Service** section, select **MO** from the **SLG Role** list.
- From the **SLG Mode** list, select **S1/S2** (SIP Proxy Server 1 and Server 2), see **Figure 8**.

Figure 8: Adding a new IP Telephony Node (cont)

- Click **Next**. The **Confirm new Node details** page appears (not shown).
- Click on the **Transfer Now** button and then the **Synchronize Configuration Files (Node ID 512)** page appears.
- Click **Finish** and wait for the configuration to be saved. The **Node Saved** page appears, see **Figure 9**.
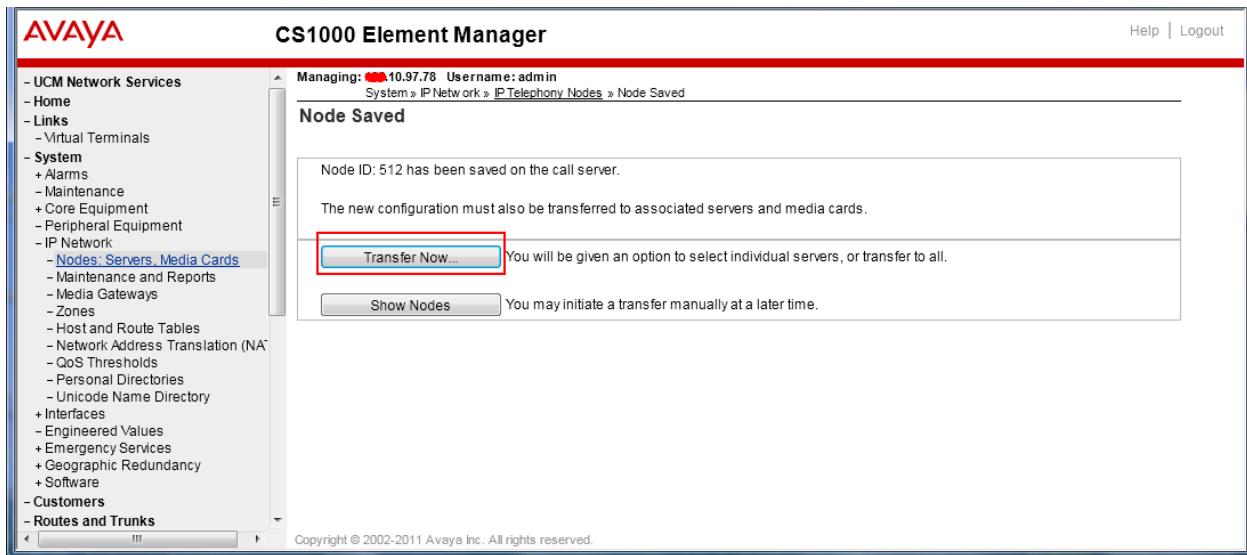
Figure 9: Node Saved with Transfer Configuration

- Select the SIP Line server that's associated with the changes and click on the **Start Sync** button to transfer the configuration files to the selected servers, see **Figure 10**.
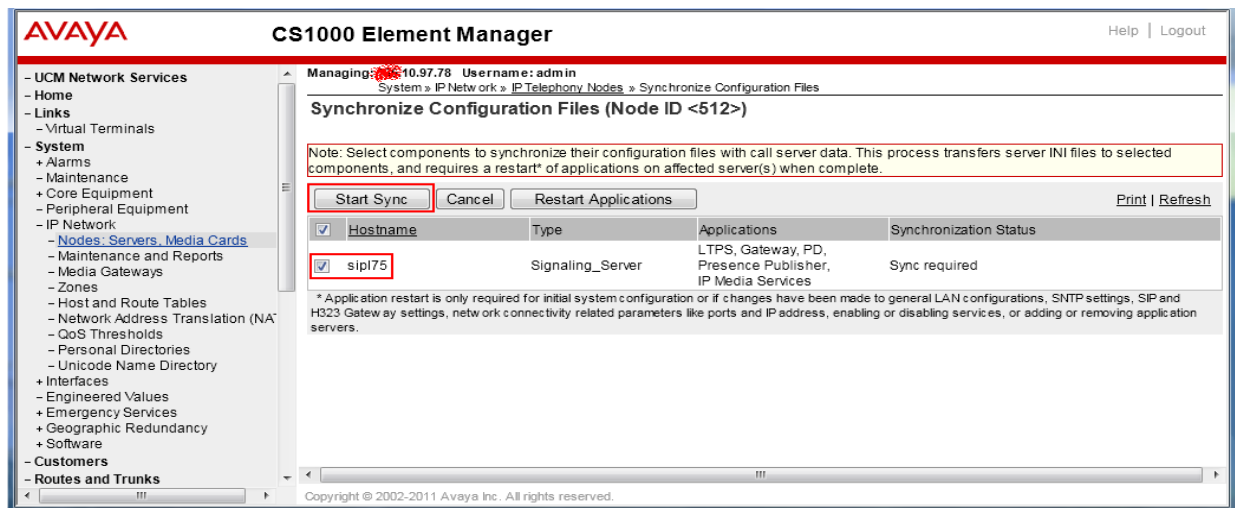

Figure 10: Synchronize Configuration Files

<u>Note</u>: The first time a new Telephony Node is added and transferred to the call server, the SIP Line services need to be restarted. To restart the SIP Line services, log in as administrator to the command line interface of the SIP Line server and issue the command: **appstart restart**.

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5.5. Create a D-Channel for SIP Line

On the EM page, on the left column menu navigate to **Routes and Trunks -> D-Channels**.
Under the **Configuration** section as shown in **Figure 11**, enter a number in the **Choose a D-Channel Number** field, and click on the **to Add** button.
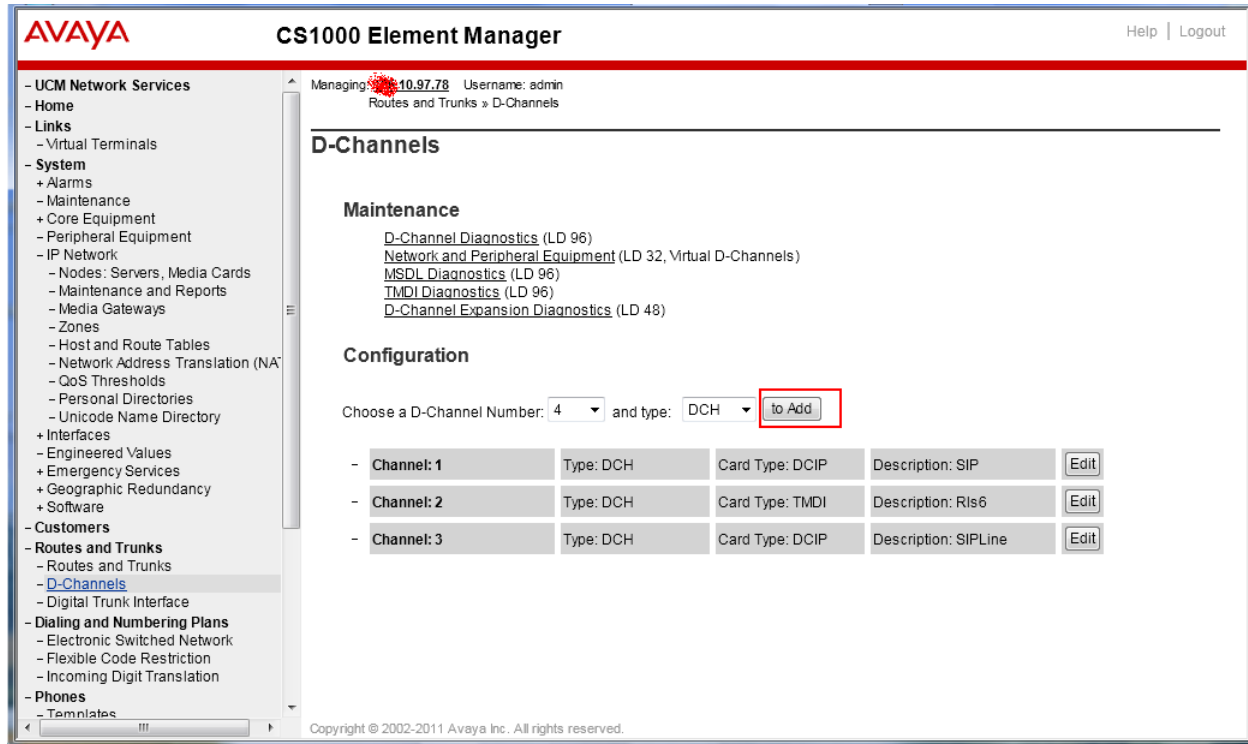

Figure 11: D-Channels configuration page

- The **D-Channels xx Property Configuration** page appears as shown in **Figure 12**.
- From the **Interface type for D-channel (IFC)** list, select **Meridian Meridian1 (SL1)**.
- Leave the other fields at default values.

Figure 12: SIP Line D-Channel Property Configuration

- Click on the **Basic options (BSCOPT)** link. The **Basic options (BSCOPT)** list expands (not shown).
- Click on **Edit** to configure **Remote Capabilities (RCAP)**. The **Remote Capabilities Configuration detail page** will appear as shown in **Figure 13**.
- Select the **Message waiting interworking with DMS-100 (MWI***)* check box.
- Select the **Network name display method 2 (ND2)** check box.
- At the bottom of the **Remote Capabilities Configuration** page, click **Return - Remote Capabilities** to return to the **D-Channel xx Property Configuration** page.
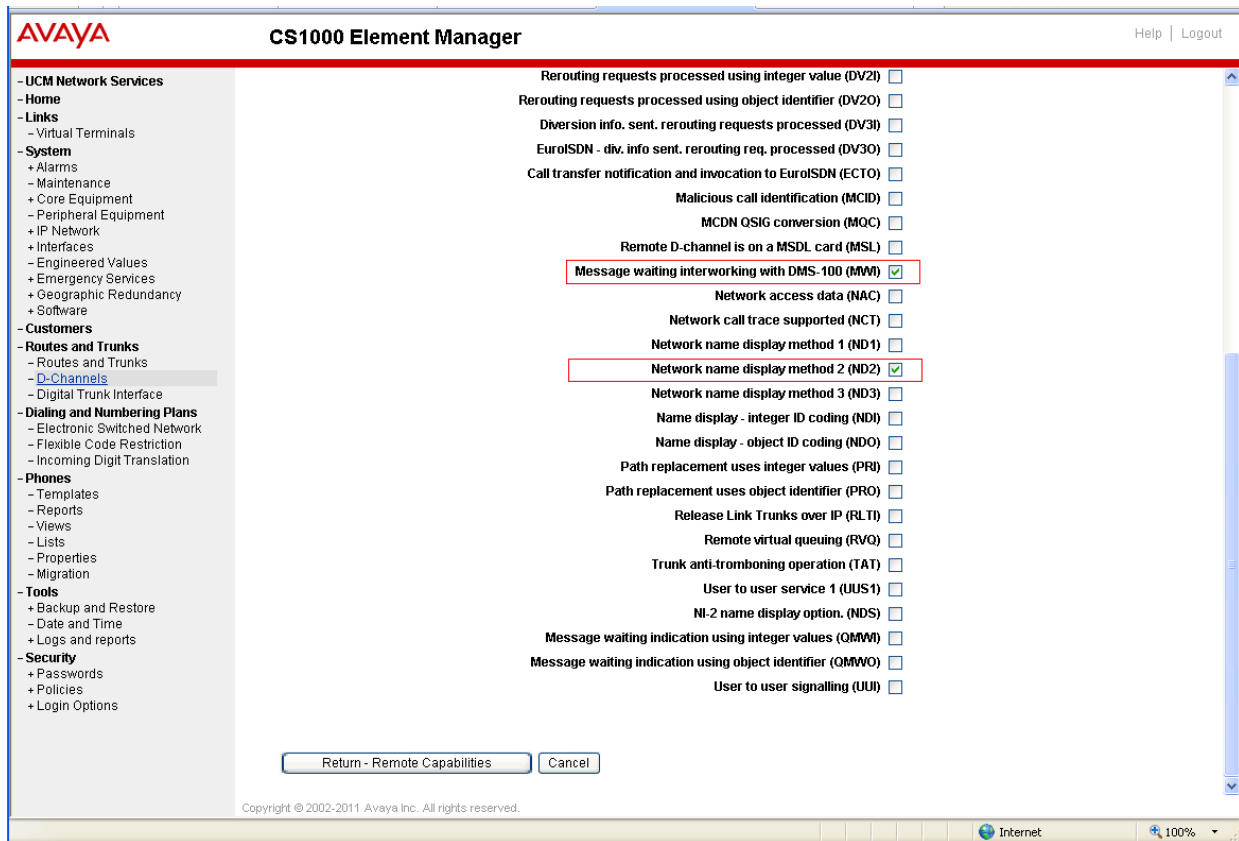
Figure 13: SIP Line D-Channel RCAP Configuration Details

- **Message Waiting Interworking with DMS-100 (MWI)** must be enabled to support voice mail notification on SIP Line endpoints.
- **Network Name Display Method 2 (ND2)** must be enabled to support name display between SIP Line endpoints.
- Other check boxes are left unchecked.

Click on the **Submit** button of the D-Channel Property Configuration page to save changes.

5.6. Create an Application Module Link (AML)

On the EM page, navigate to **System -> Interfaces -> Application Module Link**, click on the **Add** button to add a new Application Module Link (not shown). The **New Application Module Link** page appears as shown in **Figure 14**.

Enter an AML port number in the **Port number** text box. The AML for SIP Line Service can use any port from 32 to 127. In this case, SIP Line Service is configured to use port 33.

Click on the **Save** button to complete adding the AML link, and to save the configuration.
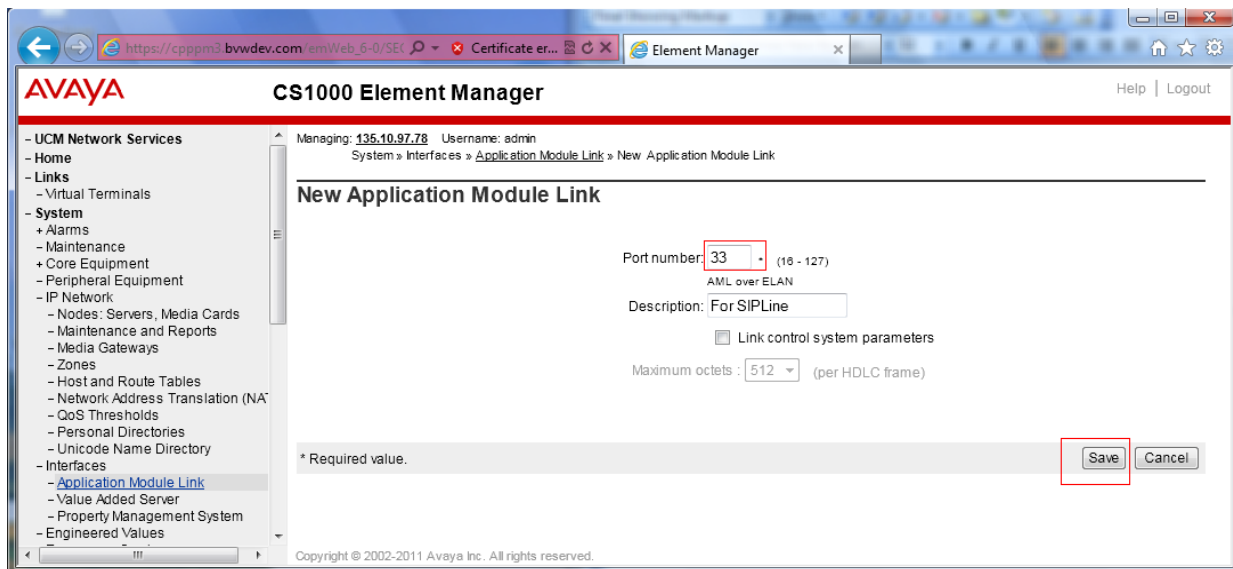
Figure 14: Adding a new AML

5.7. Create a Value Added Server (VAS)

On the EM page, navigate to **System -> Interfaces -> Value Added Server** and click on the **Add** button to add a new VAS.

The **Value Added Server** page appears (not shown), on this page, select **Ethernet Link** and the **Ethernet Link** page appears as shown in **Figure 15**.

Enter a number in the **Value added server ID** field, in this example **33** was used. In the **Ethernet LAN Link** drop down list, select the AML number of the ELAN that was created in **Section 5.6**.

Leave other fields as default values and click on the **Save** button to complete adding the **VAS,** and to save the configuration.

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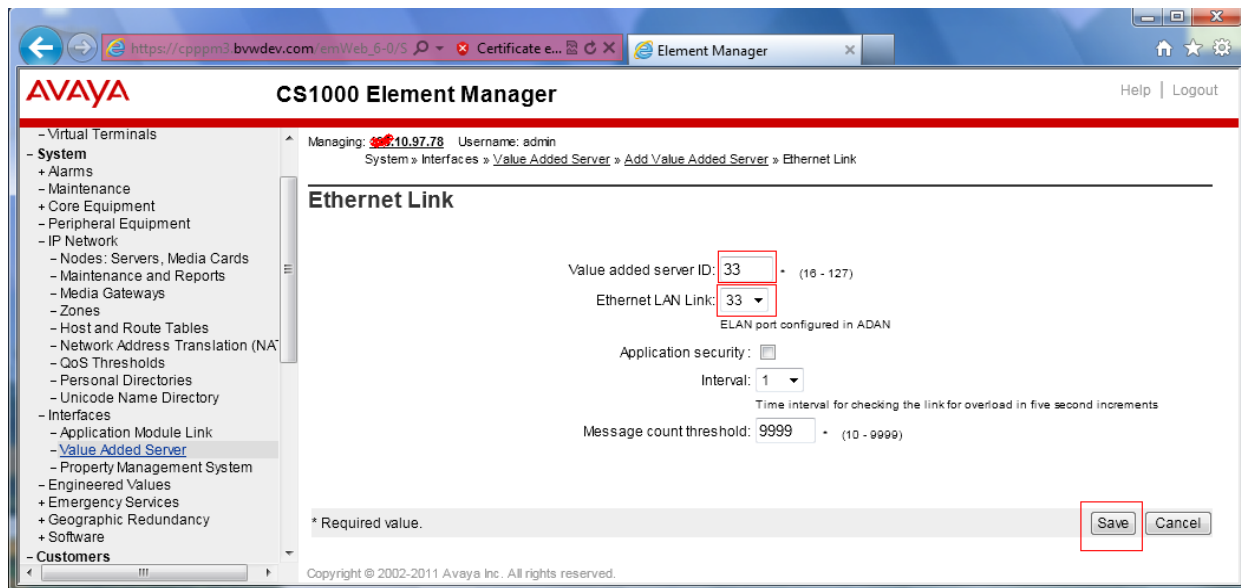
Figure 15: Adding a new Value Added Service for the AML

5.8. Create a Virtual Trunk Zone

On the EM page, navigate to menu **System -> IP Network -> Zones**. The **Zones** page appears on the right, in this page select the **Bandwidth Zones** link (not shown).

On the **Bandwidth Zones** page, click on the **Add** button (not shown), the **Zone Basic Property and Bandwidth Management** page appears as shown in **Figure 16**.

Enter a zone number in the **Zone Number (Zone)** field and in the **Zone Intent (ZBRN)** drop down menu select **VTRK (VTRK)**.

Leave other fields as default values and click on the **Save** button to complete adding the Zone.

Note: Repeat the step above to create another zone for the SIP Line phone; however remember to select **MO**, instead of VTRK in the field **Zone Intent**.

Figure 16: Adding a new Zone for Virtual Trunk

5.9. Create a SIP Line Route Data Block (RDB)

On the EM page, navigate to the menu **Routes and Trunks -> Routes and Trunks**; the **Routes and Trunks** page appears (not shown). On this page, click on the **Add route** button next to the customer number that the route will belong to.

The **Customer ID, New Route Configuration** page appears, expand the **Basic Configuration** tab, and enter values below and as shown in **Figure 17** and **18**.
- **Route Number (ROUT)**: 3
- **Trunk type(TKTP)**: TIE
- **Incoming and Outgoing trunk (ICOG)**: Incoming and Outgoing (IAO)
- **Access Code for Trunk group (ACOD)**: enter a number for ACOD, for example 7575.
- **The route is for a virtual trunk route (VTRK)**: Checked.
- **Zone for codec selection and bandwidth management (ZONE***)*: 4, this is the Virtual trunk zone number that was created in **Section 5.8**.
- **Node ID of signaling server of this route (NODE)**: 512, this is the node ID of the SIP Line.
- **Protocol ID for the route (PCID)**: SIP Line (SIPL).
- **Integrated services digital network option (ISDN)**: checked.
- **Mode of operation (MODE):** Route uses ISDN Signaling Link (ISLD).
- **D channel number (DCH)**: 3 the D-channel number that was created in the **Section 5.5**.
- **Interface type for route (IFC)**: Meridian M1 (SL1).
- **Network calling name allowed (NCNA)**: checked.
- **Channel type (CHTP)**: B-channel (BCH).
- **Call type for outgoing direct dialed TIE route (CTYP)**: CDP.
- **Calling Number dialing plan (CNDP)**: CDP.

Leave default values for The **Basic Route Options, Network Options, General Options, and Advanced Configurations** sections.

Click the **Submit** button (not shown) to complete adding the route and to save the configuration.

Figure 117: SIP Line Route Configuration

Figure 18: SIP Line Route Configuration (cont)

5.10. Create SIP Line Virtual Trunks

On the EM page, navigate to **Routes and Trunks -> Routes and Trunks** and select the **Add route** button beside the route that was created in the **Section 5.9** above to create new trunks.

The **Customer ID, Route ID, Trunk type TIE trunk data block** page appears as shown in **Figure 19**, enter values for fields as shown below:
- **Multiple trunk input number (MTINPUT)**: 32 **->** create 32 trunks.
- **Auto increment member number**: checked.
- **Trunk data block (TYPE)**: IP Trunk (IPTI).
- **Terminal Number (TN)**: 100 0 2 0 **->** enter the first TN of a range TN.
- **Member number**: 33, this is the ID of the trunk, just enter the first ID for the first trunk, next ID will be automatically created and incremented.
- **Start arrangement Incoming**: Immediate (IMM).
- **Start arrangement Outgoing**: Immediate (IMM).
- **Trunk Group Access Restriction (TGAR)**: 1.
- **Channel ID for this trunk**: 33, this ID should be the same as the ID of the Member Number.

Click on the **Class of Service** button and assign following class of services (not shown):
- **Media security**: Media Security Never (MSNV).
- **Restriction level**: Unrestricted.

Leave other fields at their default values and click on the **Return Class of Service** button (not shown) to return to the **Trunk type TIE trunk data block** page**.**

Click on the **Save** button to complete adding virtual trunks for the SIP Line.

Figure 19: Adding virtual trunks for SIP Line Trunk

5.11. Create a SIP Line Phone

To create a SIP Line phone on the Call Server, log in as administrator using the command line interface (CLI) and issue the overlay (LD) 11/20 as shown below.

The bold fields must be properly inputted as they are configured on the Call server, for other fields hit enter to leave it at default values.

```
LD20
PT0000
REQ: new
TYPE: UEXT -> Universal extension type for SIP Line phone
TN  104 0 0 1

DES  POLY1 -> Description of Phone.
CUST 0
```

UXTY SIPL -> Universal extension type is SIP Line
MCCL YES
SIPN 0
SIP3 1 -> For SIP phone third party, enter 1 in this field
FMCL
TLSV
SIPU 54008 -> SIP phone username
NDID 512 -> Node ID of SIP Line
SUPR
SUBR
UXID
NUID
NHTN
ZONE 3 -> Zone for SIP Line phone.
MRT
ERL
ECL
VSIT
FDN 54002 -> Forward No Answer to this DN, need to enable class of service FNA
TGAR 1
LDN
NCOS 7 -> Network Class of Service, 7 is highest level.
SGRP
RNPG
SCI
SSU
XLST
SCPW 1234 → Password to log in to SIP Line username 54008
SFLT
CAC_MFC
CLS FNA FBA HTA MWA DNDA CNDA CFXA -> class of service.
RCO
HUNT 54444 -> Forward busy to this DN, need to enable class of service FBA and HTA
PLEV
KEY 00 SCR 54008 0   MARP -> Key 0 is DN of SIP phone.
    CPND new
     CPND_LANG ROMAN
      NAME Poly 8440 -> Display name of SIP Phone.
      XPLN 13
      DISPLAY_FMT FIRST,LAST
  01 HOT U 2654008 MARP 0 -> Key 1 Hot U with prefix + DN
  02 CWT -> Call Waiting key
  03 MSB -> Make Set busy key
  04 SCU 0000 -> Speech call dial key

6. Configuring the Algo 8028

This section explains the steps required to configure the Algo 8028 to interoperate with CS1000. The assumption is made that all the required wiring between the Algo 8028 controller and the Door-station is successfully completed. For complete information on 8028 installation and configuration refer to **Section 9[2]**.

6.1. Obtaining the IP Address of the Doorphone Control Panel

The Algo 8028 controller has an Ethernet and Door Station jack. During compliance testing the Ethernet jack was connected to a PoE switch in the lab providing IP addresses through a DHCP server. Connect the Door Station jack to the Algo 8028 Door Station. To obtain the IP Address (DHCP) of the Algo 8028 controller, press the call button on the Door Station. A recorded voice will speak the IP address that was assigned to the Algo 8028 controller. During compliance testing *10.10.98.20* was the assigned IP address by the DHCP server.

6.2. Configuring the 8028 SIP Doorphone Control Panel

Open a browser and enter the IP address of the 8028 controller in the URL. The IP address was obtained as explained in **Section 6.1**. The Welcome screen of the Algo 8028 SIP Doorphone Control Panel is presented as shown in **Figure 20**. Enter the **Password** and click on **Login**. The default password is **algo**.

Figure 20: Doorphone Control Panel Login Screen

Navigate to the **Config** tab as shown in **Figure 21**. Under the **SIP** section, enter the **SIP Domain/Proxy** value as highlighted in the red box. Enter the value of the **Extension** along with the **Auth ID**, **Password** and **Dialing Extension**. This is the Extension that will ring when the user presses the call button of the Door Station. During compliance testing extension *54009* was used. All other values are left at default.

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Figure 21: Configuring the SIP Doorphone Control Panel

At the **Features** section, all the parameters can be left as default. Click **Save Settings** to complete the configuration as show in **Figure 22**.

Figure 22: Configuring the SIP Doorphone Control Panel (Continue)

Figure 23 shows the **Status** of the Algo 8028 controller where the Extension *54009* is successfully registered to the CS1000 as a SIP endpoint.

Figure 23: Configuring the SIP Doorphone Control Panel

7. Verification Steps

The following tests were conducted to verify the solution between the Algo 8028 and Avaya CS1000.

- Verify that the Algo 8028 controller registers as a SIP endpoint with CS1000.
- Verify that when the call button on the Door Station is pressed the telephone on the CS1000 rings and a clear speech path is established.
- Verify that the telephone that receives the incoming call from the Door Station can do conference, transfer, mute, un-mute and provide busy tone if it is on another call.
- Verify that the solution works on various Avaya telephones and that DTMF tones from all these different telephone types works with the Algo 8028 controller by unlocking the door release.
- Verify that Algo 8028 goes into an idle state when the call is completed.
- Verify that the Algo 8028 re-registers without issues if the Ethernet cable is unplugged and plugged back in.

8. Conclusion

All of the executed test cases have passed and met the objectives outlined in the **Section 2.1**, with some exceptions outlined in **Section 2.2**. The Algo 8028 firmware version 1.1.4 is considered to be in compliance with Avaya CS1000 SIP Line System Release 7.5.

9. Additional References

[1] Product documentation for the Avaya CS1000 products may be found at:
https://support.avaya.com/css/Products/

Avaya Communication Server 1000E Installation and Commissioning
Avaya Communication Server 1000 SIP Line Fundamental, Release 7.5
Avaya Communication Server 1000 Element Manager System Reference – Administration
Avaya Communication Server 1000 Co-resident Call Server and Signaling Server
Fundamentals
Avaya Communication Server 1000 Unified Communications Management Common Services
Fundamentals
 Avaya Communication Server 1000 ISDN Primary Rate Interface Installation and
Commissioning

[2] Product documentation for the Algo 8028 products may be found at:
www.algosolutions.com/8028