

Application Notes for Ascom Wifi i62 SIP Telephone firmware version 2.2.22 with Avaya Communication Server 1000 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 Ascom Wifi i62 SIP telephone. During the compliance testing, the Ascom Wifi i62 was able to register as a SIP client endpoint with Communication Server 1000 SIP Line gateway. The Ascom Wifi i62 telephone was able to place and receive calls from Communication Server 1000 Release 7.5 non-SIP and SIP Line telephones. The compliance testing focused on basic telephone features.

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 Release 7.5 (hereafter referred to as CS 1000) and the Ascom Wifi i62 SIP telephone firmware version 2.2.22 used during the compliance testing. The Ascom Wifi i62 was tested with non-SIP and SIP telephones using the CS1000 SIP line Release 7.5. All the applicable telephony feature test cases of Release 7.5 SIP line were executed on the Ascom Wifi i62, where applicable, to ensure that the interoperability with CS 1000.

2. General Test Approach and Test Results

The general test approach was to have the Ascom Wifi i62 telephone to register to the CS1000 SIP line gateway. Calls were then placed from other CS1000 telephone clients/users to and from the Ascom Wifi i62 telephone. Other telephony features such as busy, hold, DTMF, MWI and codec negotiation were also verified.

2.1. Interoperability Compliance Testing

Avaya's formal testing and Declaration of Conformity is provided only on the headsets/handsets that carry the Avaya brand or logo. Avaya may conduct testing of non-Avaya headset/handset to determine interoperability with Avaya phones. However, Avaya does not conduct the testing of non-Avaya headsets/handsets for: Acoustic Pressure, Safety, Hearing Aid Compliance, EMC regulations, or any other tests to ensure conformity with safety, audio quality, long-term reliability or any regulation requirements. As a result, Avaya makes no representations whether a particular non-Avaya headset will work with Avaya's telephones or with a different generation of the same Avaya telephone.

Since there is no industry standard for handset interfaces, different manufacturers utilize different handset/headset interfaces with their telephones. Therefore, any claim made by a headset vendor that its product is compatible with Avaya telephones does not equate to a guarantee that the headset will provide adequate safety protection or audio quality.

The focus of this testing was to verify that the Ascom Wifi i62 SIP telephone was able to interoperate with the CS 1000 SIP line system. The following areas were tested:

- Registration of the Ascom Wifi i62 SIP telephone to the CS1000 SIP Line Gateway.
- Call establishment of Ascom Wifi i62 with CS1000 SIP and non-SIP telephones.
- Telephony features: Basic calls, conference, blind and consultative transfer, DTMF (dual tone multi frequency) RFC2833 and SIP Info transmission, voicemail with Message Waiting Indication (MWI) notification, busy, hold, speed dial, group call pickup, call park, call waiting, ring again busy/no answer, multiple appearances Directory Number and Call forward on Busy, No answer and All Calls..
- PSTN calls over PRI trunk.
- Codec negotiation G.711 and G.729.

2.2. Test Results

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

- 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 PSTN over a PRI trunk.
- Local Call Waiting and Call Forward Busy are not support due to CS1000 SIP Line Gateway will always response with 486 Busy Here.

2.3. Support

Technical support for the Ascom IP DECT product can be obtained through a local Ascom supplier. Ascom global technical support:

• Email: <u>support@ascom.se</u> or Help desk: +46 31 559450

3. Reference Configuration

Figure 1 illustrates the test configuration used during the compliance testing between the Avaya CS1000 and the Ascom Wifi i62.



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	3.04.0003
Avaya IP Phone 1140	0625C6O
Avaya IP Phone 2004P2	0692D93
Avaya IP Phone 2002P2	0604DC5
Avaya SIP 1140	02.02.21.00
Avaya Aura® Session Manager	6.1
Ascom Communication equipment	WIFI i62 sets firmware version 2.2.22
	Wireless Access Point
	WinPDM 3.8.1 (Device Manger for Windows)

5. Configure Avaya CS 1000 - SIP LINE

This section describes the steps to configure the Avaya CS1000 SIP Line using CS 1000 Element Manager. A command line interface (CLI) option is available to provision the SIP Line application on the CS 1000 system. For detailed information on how to configure and administer the CS 1000 SIP Line, please refer to **Section 9 Reference [1]**.

The following is the summary of tasks needed to be done for configuring the CS 1000 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 CS 1000 Release 7.5 Linux Base.
- Joined CS 1000 Release 7.5 Security Domain.
- 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 CS 1000 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.

Αναγα	Avaya Unified Communica	ations Manageme	ent		Help Logout
- Network Elements	Host Name: car2-sipl-ucm.bwdev.com	Software Version: 02.20-SN	APSHOT(0000) User Name admin	I	
- CS 1000 Services IPSec	Elements				
Patches SNMP Profiles Secure FTP Token	New elements are registered into the secur can optionally filter the list by entering a sea	ity framework, or may be add rch term.	led as simple hyperlinks. Click an ele	ment name to launch its manager	nent service. You
Software Deployment — User Services	Sea	arch Reset			
Administrative Users External Authentication	Add Edit Delete				⊕ <u>¤</u> ≣
Password	Element Name	Element Type +	Release	Address	Description 🔺
Roles	1 EM on car2-cores	CS1000	7.5		New element.
Policies Certificates	2 EM on car2-ssq-carrier	CS1000	7.5		New element.
Active Sessions — Tools	3 EM on cpppm3	CS1000	7.5		New element =
Logs	4 car2-ssq-carrier.bwdev.com (member)	Linux Base	7.5		Base OS element
Data	5 car2-sipl-ucm.bvwdev.com	Linux Base	7.5	************** *	Base OS element
	e car2-mas.bwdev.com (member)	Linux Base	7.5		Base OS element
	7 ar2-cores.bwdev.com (member)	Linux Base	7.5	****	Base OS element
	8 car2-sps.bwdev.com (member)	Linux Base	7.5		Base OS element.
	9 cpppm3.bwdev.com (member)	Linux Base	7.5		Base OS element
	im sini75 hwwdav.com (mamhar)	Linuv Roea	7.5	Constant Statements	Passa OQ *
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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 CS 1000 system that needs to be configured, in this sample that is **cpppm3**. The CS 1000 Element Manager page appears as shown in **Figure 3** below.

Αναγα	CS1000 Element Manager	Help Logout
- UCM Network Services - Home - Links - Virtual Terminals	Managing: With 0.97.78 Username: admin System Overview	
 - System + Alarms - Maintenance + Core Equipment 		_
- Peripheral Equipment + IP Network + Interfaces - Engineered Values + Emergency Services	IP Address: 10.10.97.78 Type: Awaya Communication Server 1000E CPPM Linux Version: 4121 Release: 75.00 +	
+ Geographic Redundancy + Software - Customers - Routes and Trunks Boutes and Trunks		
- D-Channels - Digital Trunk Interface - Digital Trunk Interface - Dialing and Numbering Plans - Electronic Switched Network		
 Flexible Code Restriction Incoming Digit Translation Phones Templates 		
- Reports - Views - Lists - Properties - Mioration		
- Tools + Backup and Restore - Date and Time + Logs and reports		
- Security + Passwords + Policies + Login Options		
	Copyright © 2002-2011 Avaya Inc. All rights reserved.	

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; 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.

Αναγα	CS1000 Element Manager	Help Logout
- UCM Network Services - Home - Links - Mrtual Terminals - System + Alarms	Managing: <u>diffe 10.97.78</u> Username: admin <u>Oustomers</u> » Oustomer 00 » <u>Oustomer Details</u> » SIP Line Servic e SIP Line Service	
- Maintenance + Core Equipment - Peripheral Equipment + IP Network + Interfaces - Engineered Values + Emergency Services	 ✓ SIP Line Service User agent DN prefix: 26 Optional features: ✓ Nortel Multimedia 	
	"Required Value	Save Cancel
	Copyright © 2002-2011 Avaya Inc. All rights reserved.	

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 \rightarrow IP Network \rightarrow Nodes: Servers, Media Cards. Click Add to add a new SIP Line Node to the IP Telephony Nodes. 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 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 SIP endpoint uses to register to.
- Subnet Mask text box: 255.255.255.192.
- Embedded LAN (ELAN) Gateway IP Address text box: 10.10.97.66.
- Embedded LAN (ELAN) Subnet Mask text box: 255.255.255.192.
- Check **SIP Line** check box to enable SIP Line for this Node.

- UCM Network Services - Managing:	
- Links New IP Telephony Node - Mirtual Terminals Step 1: Define the new Node and its services.	
- Virtual Terminals Step 1: Define the new Node and its services.	
- System	
+ Alarms	
- Maintenance	
+ Core Equipment	
- Peripheral Equipment Node ID: 512 * (0-9999)	
- IP Network	
- Nodes: Servers, Media Cards Call server IP address: 10.10.97.78 * TLAN address type: IPv4 only	
- Maintenance and Reports	
- media Gateways	
- Loies	
Network Address Translation (NA: Embedded LAN (ELAN) Telephony LAN (TLAN)	
- QoS Thresholds = Gateway IP address: 10.10.97.65 * Node IPv4 address: 10.10.97.187 *	
- Personal Directories	
- Unicode Name Directory Subnet mask: 255 255 255 192 * Subnet mask: 255 255 255 192 *	
+ Interfaces	
- Engineered Values	
+ Emergency Services	
+ Geographic Kedundancy	
+ JOURATE Annications: SILL in a	
- Customers Applications. To child the Application of the Control of Table 1	
Ovisitim Line Terminal Proxy Server (LTPS)	
- Duchannels Virtual Trunk Gateway (SIPGw, H323Gw)	
Digital Trunk Interface Personal Directory (PD)	
- Dialing and Numbering Plans	
- Electronic Switched Network	
- Flexible Code Restriction * Required Value. Next > Cancel	
- Incoming Digit Translation	
- Phones	
- Templates	
- Keports	
- views	
- Dronordies	
- Migration	
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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).

Αναγα	c	S1000 Element Manager	Help Logout
- UCM Network Services - Home - Links - Virtual Terminals - System + Alarms - Maintenance + Core Equipment		Managing: 10.97.78 Username:admin System »: IP Network »: IP Telephony Nodes » New IP Telephony Node New IP Telephony Node (ID:513) 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.	-
– Peripheral Equipment – IP Network	L	Select to add Add Remove Make Leader Print Refresh	
- Nodes: Servers, Media Cards - Maintenance and Reports		Hostname Type Deployed Applications ELAN IP TLAN IPv4 TLAN IPv6 Role	
- Zones - Aost and Route Tables - Network Address Translation (NA' - QoS Thresholds - Personal Directories - Unicode Name Directory Interfaces - Engineered Values + Emergency Services + Geographic Redundancy + Software - Customers - Routes and Trunks - Routes and Trunks - Digital Trunk Interface - Digital Trunk Interface - Digital Trunk Interface	в	Select from the list above and click Add to associate servers with this node. Selected servers must have identical application deployments.	
- Electronic Switched Network - Flexible Code Restriction - Incoming Digit Translation		Kenter State St]
Phones Templates - Reports - Views Lists Properties - Migration finite f	•	∑opyright © 2002-2011 Avaya Inc. All rights reserved.	

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

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

Αναγα	CS1000 Element Manager	Logout
UCM Network Services Home Links Virtual Terminals System Alarms Maintenance Core Equipment Peripheral Equipment IR Network	Managing:	
- Nodes: Servers, Media Cards	SIP domain name: sigl75.com Monitor IP addresses (listed below)	
 Media Gateways Zones Host and Route Tables Network Address Translation (NA) 	SLG endpoint name: sipline Information will be captured for the IP addresses listed below.	
- Rework Address Interstation (WA - QoS Thresholds - Personal Directories - Unicode Name Directory Interfaces	SLG Group ID: Monitor IP: Add SLG Local Sip port: 5070 (1 - 65535)	
Engineered Values Emergency Services Geographic Redundancy	SLG Local TIs port 5071 (1 - 65535)	
+ Sonware - Customers - Routes and Trunks - Routes and Trunks - D-Channels - Dicital Trunk Interface	Security policy: Security Disabled Number of byte re-negotiation: 0 Options: Client authentication	
Digital ministration and Numbering Plans Electronic Switched Network Flexible Code Restriction Incoming Digits Translation	* Required Value. Note: Changes made on this page will NOT be Save Cancel transmitted until the Node is also saved.	
- Templates - Templates - Reports - Usis - Properties - Properties - Maration - Maratio	⊂ Copyright © 2002-2011 Avaya Inc. All rights reserved.	

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.

Αναγα	CS1000 Element Manager	Help Logout
- UCM Network Services	Managing: 📫 10.97.78 Username: admin	
- Home	System » IP Network » IP Telephony Nodes » Node Details » SIP Line Configuration	
- Links	Node ID: 512 - SIP Line Configuration Details	
- Virtual Terminals		
- System	Operated L OID Line Ophysics Dollars Colours Decision	
+ Alarms	General SIP Line Gateway Settings SIP Line Gateway Service	
- Maintenance	SIP Line Gateway Service	
+ Core Equipment	Branch / GD Office Settings	
- Peripheral Equipment		
- IP Network	SLG role: MO 🗸	
- Nodes: Servers, Media Cards	SLG mode: S1/S2 -	
- Maintenance and Reports		
- Media Gateways	MO SLG IPv4 address: 0.0.00	
- Host and Route Tables	The IP address can have either IPv4 or IPv6 format based on the value of "TLAN	
- Network Address Translation (NA*	address type"	
- QoS Thresholds	MO SLG IPI6 address:	
- Personal Directories	NO SLO IF VO AUGESS.	
- Unicode Name Directory	MO SL G port 5070 (1 - 65535)	
+ Interfaces		
 Engineered Values 	MO SLG transport TCP -	
+ Emergency Services		
+ Geographic Redundancy	GR SLG IPv4 address: 0.0.0.0	
+ Software	The IP address can have either IPv4 or IPv6 format based on the value of "TLAN	
- Customers	address type"	
- Routes and Trunks	GR SLG IPV6 address:	
- Routes and Trunks		
- D-Channels Digital Trupk Interface	GR SLG port 5070 (1 - 65535)	
Dialing and Numbering Diane	Note: Observe mode on this areas will NOT be	
- Electronic Switched Network	* Required Value. transmitted until the Note is also saved Save Cancel	
- Elexible Code Restriction		
- Incoming Digit Translation		
- Phones		
- Templates		
- Reports		
- Views		
- Lists		
- Properties		
- Migration	Convicient & 2022 2014 Avenue Ins. All rights responsed	
	Cupyrgin w 2002-2011 Avaya inc. All rights reserved.	

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.

Αναγα	S1000 Element Manager Help	Logout
- UCM Network Services - Home - Links - Virtual Terminals	Managing: top 10.97.78 Username:admin System » IP Network » IP Telephony Nodes » Node Saved Node Saved	_
- System + Alarms - Maintenance + Core Equipment - Peripheral Equipment - IP Network - <u>Nodes: Servers, Media Cards</u> - Maintenance and Reports	Node ID: 512 has been saved on the call server. The new configuration must also be transferred to associated servers and media cards. Transfer Now You will be given an option to select individual servers, or transfer to all.	
- Media Gateways - Zones - Host and Route Tables - Network Address Translation (NA' - GoS Thresholds - Personal Directories - Unicode Name Directory + Interfaces	Show Nodes You may initiate a transfer manually at a later time.	
- Engineered Values + Emergency Services + Geographic Redundancy + Software - Customers - Routes and Trunks <	Copyright © 2002-2011 Avaya Inc. All rights reserved.	

Figure 9: Node Saved with Transfer Configuration

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

Αναγα	cs	\$1000 Element Mana	ger		Help Logou
- UCM Network Services - Home - Links - Mitual Terminals - System	•	Managing 10.97.78 Usernan System » IP Network » I Synchronize Configura	e:admin <u>Telephony Nodes</u> » Synchro tion Files (Node ID bronize their configuration	nize Configuration Files <512>)	n mass transfore samer INI files to selected
Alarms Maintenance Horintenance Core Equipment Peripheral Equipment IP Network Nodes: Servers, Media Cards Maintenance and Reports Media Gateways Zonee		Components, and requires a res Components, and requires a res Start Sync Cancel (U Hostname U sipl75	Restart Applications on affe Restart Applications Type Signaling_Server	Applications LTPS, Gateway, PD, Presence Publisher, IP Media Services	Print Refres Synchronization Status Sync required
Host and Route Tables Network Address Translation (NA QoS Thresholds Personal Directories Unicode Name Directory Interfaces Engineered Values Emergency Services Geographic Redundancy Setwice	47	* Application restart is only required H323 Gateway settings, network co servers.	f for initial system configuratio nnectivity related parameters I	n or if changes have been made to like ports and IP address, enabling	o general LAN configurations, SNTP settings, SIP and or disabling services, or adding or removing application
+ Sonware - Customers - Routes and Trunks - III	Ŧ	✓ Copyright © 2002-2011 Avaya Inc. A	I rights reserved.	III	

Figure 10: Synchronize Configuration Files

<u>Note</u>: The first time a new Telephony Node is added and transfered 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 command: **appstart restart**.

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.

AVAYA c	S1000 Element Mai	nager			Help Logout
- UCM Network Services - Home - Links - Virtual Terminals - System - Karms - Maintenance + Core Equipment - Peripheral Equipment - IP Network - Nodes: Servers, Media Cards - Madia Gateways - Zones - Hedia Gateways - Zones - Host and Route Tables - Network Address Translation (NAT	Managing: 10.97.78 Userna Routes and Trunks » D D-Channels Maintenance D-Channel Diaon Network and Perip MSDL Diaonostics TMDI Diaonostics D-Channel Expansion Configuration	ame: admin -Channels ostics (LD 96) oheral Equipment (LD 32, Vir § (LD 96) (LD 96) sion Diagnostics (LD 48)	tual D-Channels)		
- GoS Infresholds - Personal Directories - Unicode Name Directory + Interfaces	Choose a D-Channel Nu	umber: 4 🔻 and type:	DCH 🔻 to Add		
 Engineered Values + Emergency Services 	- Channel: 1	Type: DCH	Card Type: DCIP	Description: SIP	Edit
+ Geographic Redundancy + Software	- Channel: 2	Type: DCH	Card Type: TMDI	Description: RIs6	Edit
- Customers - Routes and Trunks - Routes and Trunks	- Channel: 3	Type: DCH	Card Type: DCIP	Description: SIPLine	Edit
- <u>D-Channels</u> - Digital Trunk Interface - Dialing and Numbering Plans - Electronic Switched Network - Flexible Code Restriction - Incoming Digit Translation - Phones					
- Temniates	Copyright © 2002-2011 Avaya In	c. All rights reserved.			

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) (not shown). 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 the **D-Channel xx Property Configuration** page.

AVAYA	CS1000 Element Manager	Help Logout
- UCM Network Services	Rerouting requests processed using integer value (DV2I)	^
- Home	Rerouting requests processed using object identifier (DV20)	_
- Links	Diversion info. sent, rerouting requests processed (DV3))	
- Virtual Terminals - System	EuroISDN - div. info sent, rerouting reg. processed (DV30)	
+ Alarms	Call transfer notification and important for EuroISDN (ECCT)	
- Maintenance		
- Peripheral Equipment		
+ IP Network		
- Engineered Values		
+ Emergency Services	Message watung interworking with Dims-Tout (invol)	
+ Geographic Redundancy + Software	Network access data (NAC)	
- Customers	Network call trace supported (NCT)	
- Routes and Trunks	Network name display method 1 (ND1)	
- D-Channels	Network name display method 2 (ND2) 🔽	
– Digital Trunk Interface	Network name display method 3 (ND3)	
 Dialing and Numbering Plans Electronic Switched Network 	Name display - integer ID coding (NDI) 📃	
- Flexible Code Restriction	Name display - object ID coding (NDO)	
- Incoming Digit Translation	Path replacement uses integer values (PRI) 📃	
- Templates	Path replacement uses object identifier (PRO) 📃	
- Reports	Release Link Trunks over IP (RLTI) 📃	
- Views - Lists	Remote virtual queuing (RVQ) 🔲	
- Properties	Trunk anti-tromboning operation (TAT)	E
- Migration - Tools	User to user service 1 (UUS1)	
+ Backup and Restore	NI-2 name display option. (NDS)	
- Date and Time + Logs and reports	Message waiting indication using integer values (QMW)	
- Security	Message waiting indication using object identifier (QMWQ)	
+ Passwords	Iser to user simaling (IIII)	
+ Login Options	, (===============================	
	Return - Remote Capabilities	
	Conversite @ 2022 2014 A varia las All viete a sessional	<u> </u>
	Cupyrignt @ 2002-2011 Avaya inc. Air rights reserved.	
	Service Servic	∞ 100% ▼

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 of SIP Line Service can use a 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.

A https://coopers? braude	
C C nutps//cpppnis.bywue	
Ανάγα	CS1000 Element Manager Help Logout
UCM Network Services Home Links Virtual Terminals System + Alarms Adams A	Menaging: <u>135.10.97.78</u> Username: admin System » Interfaces » <u>Application Module Link</u> » New Application Module Link New Application Module Link Port number: <u>33</u> (18 - 127) AML over ELAN Description: For SIPLine Link control system parameters Maximum octets : <u>512</u> (per HDLC frame)
- Interfaces	* Required value. Save Cancel
- <u>Application Module Link</u> - Value Added Server - Property Management System - Engineered Values	•
 ✓ ✓ 	Copyright © 2002-2011 Avaya Inc. All rights reserved.

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), in this page, select the Ethernet Link 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 ELAN that was created in the Section 5.6.

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

A ttps://cpppm3.bvwd	ev.com/emWeb_6-0/S 🔎 👻 Certificate e 🗟 🕈 🗙 🥔 Element Manager 🛛 🖌 🏠
Αναγα	CS1000 Element Manager Help Logout
- Virtual Terminals - System + Alarms - Maintenance + Core Equipment - Peripheral 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 - Application Module Link - Value Added Server - Property Management System - Engineered Values	Menaging: self:10.97.78 Username: admin System > Interfaces > Value Added Server > Add Value Added Server > Ethernet Link Ethernet Link Image: System > Interfaces > Value Added Server > Add Value Added Server > Ethernet Link Image: System > Interfaces > Value Added Server > Add Value Added Server > Ethernet Link Image: System > Interfaces > Value Added Server > Add Value Added Server > Ethernet Link Image: System > Interfaces > Value Added Server D: 33 + (16 - 127) Image: System > Interfaces > Value Added Server D: 33 + (16 - 127) Image: System > Interfaces > Value Added Server D: 33 + (16 - 127) Image: System > Interfaces > Value Added Server D: 33 + (16 - 127) Image: System > Interfaces > Value Added Server D: 33 + (16 - 127) Image: System > Interfaces > Value Added Server D: 33 + (16 - 127) Image: System > Interfaces > Value Added Server D: 33 + (16 - 127) Image: System > Interfaces > Value Added Server D: 33 + (16 - 127) Image: System > Interfaces > Value Added Server D: 33 + (16 - 127) Image: System > Interfaces > Value Added Server D: System > (10 - 9999) Image: System > Interfaces > Value Added Server D: System > (10 - 9999)
+ Geographic Redundancy + Software - Customers	* Required value.

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 **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.

<u>Note</u>: 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**.

Αναγα	CS1000 Element Manager	lelp Logout
UCM Network Services Home Links Virtual Terminals System Aarms Maintenance Core Equipment Peripheral Equipment IP Network Nodes: Servers, Media Cards Maintenance and Reports Maintenance and Reports Media Gateways Zones Host and Route Tables Network Address Translation (NA CoS Thresholds Personal Directories Unicode Name Directory Interfaces Aplication Module Link Value Added Server	Managing: #£.10.97.78 Username: admin System » IP Network » Zones » Bandwidth Zones » Zone Basic Property and Bandwidth Management Zone Basic Property and Bandwidth Management Zone Number (ZONE): 4 • (1-8000) Intrazone Bandwidth (INTRA_BW): 1000000 (0 - 10000000) Intrazone Strategy (INTRA_STGY): Best Quality (BQ) • Interzone Bandwidth (INTER_BW): 1000000 (0 - 10000000) Interzone Strategy (INTER_STGY): Best Quality (BQ) • Resource Type (RES_TYPE): Shared (SHARED) • Zone Intent (ZBRN): MO (MO) • Description (ZDES): •	
- Engineered Values + Emergency Services	* Required value. Save	Cancel
+ Geographic Redundancy + Software	Copyright © 2002-2011 Avaya Inc. All rights reserved.	-

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). In 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): 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 created in the 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): 4, 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 to complete adding the route and save configuration.







Figure 18: SIP Line Route Configuration (cont)

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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, and 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 ID of trunk, just enter the first ID for 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 with the ID of 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 default values and click on the **Return Class of Service** button to return to the **Trunk type TIE trunk data block** page.

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

AVAYA	CS1000 Element Manager	Help Logout
- UCM Network Services - Home - Links	Managing: <u>Asse 10.97.78</u> Username: admin Routes and Trunks » <u>Routes and Trunks</u> » Customer 0, Route 3	
- Virtual Terminals - System + Alarms - Maintenance	Customer 0, Route 3, Frunk type TIE trunk o	data block
+ Core Equipment - Peripheral Equipment + IP Network	- Basic Configuration Multiple trunk input number:	32 Range: 2 - 3700
- Engineered Values + Emergency Services + Geographic Redundancy	Auto increment member number. Trunk data block: Terminal number:	IP Trunk (IPTI) ▼ 100 0 2 0 *
+ Software - Customers - Routes and Trunks	Designator field for trunk: Extended trunk:	SIPLINE
- D-Channels - Digital Trunk Interface - Dialing and Numbering Plans	Member number: Level 3 Signaling:	33 *
 Electronic Switched Network Flexible Code Restriction Incoming Digit Translation 	Card density: Start arrangement Incoming :	Octal Density (8D) V Immediate (IMM) V
- Templates - Reports - Views	Start arrangement Outgoing: Trunk group access restriction:	Immediate (IMM)
- Lists - Properties - Migration	Channel ID for this trunk: Class of Service:	33 Edit
 Hools Backup and Restore Date and Time Logs and reports 	+ Advanced Trunk Configurations * Required value.	Save Cancel -
- Security	Copyright © 2002-2011 Avaya Inc. All rights reserved.	

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 **104001** 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 usemame 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 FMTFIRST,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. Configure Ascom Wifi i62

This section describes how to access and configure the Ascom Wifi i62 SIP handset via the Windows Device Manager called WinPDM version 3.8.1, which can be downloaded via Ascom extranet and installed on a Windows PC. Remote device management "over the air" provides a similar graphical user interface. Insert the handset to be configured in the DP1 USB cradle, start the Ascom Device Manager, and select the "Devices" tab. The inserted i62 set is now being indicated with a check mark under the **Online** column as shown in **Figure 20**.

File Device No	umber Template	License Options	Help					
Devices Numbe	ers Templates Lice	nses						
DX D								
Delete Ungrade	software Cancel							
perete opgrade								
Device types:	Conrela fore		in: Device I					
Device types:	Search for:	20	III. Device I					
(All)	Device ID	Device type	Software version	Parameter version	Upgrade status	Online	Latest number	
(All) 62 Messenger	Device ID 00013E1218E2	Device type i62 Talker	Software version	Parameter version	Upgrade status	Online	Latest number	
(All) (62 Messenger (62 Talker	Device ID 00013E1218E2 00013E121938	Device type i62 Talker i62 Talker	Software version 2.2.22 2.2.22	Parameter version 13.16 13.16	Upgrade status	Online	Latest number 54009	
(All) (62 Messenger (62 Talker	Device ID 00013E1218E2 00013E121938 00013E122683	Device type i62 Talker i62 Talker i62 Messenger	Software version 2.2.22 2.2.22 2.2.17	Parameter version 13.16 13.16 13.16 13.16	Upgrade status	Online ✓	Latest number 54009 54004	
(All) 62 Messenger 62 Talker	Device ID 00013E1218E2 00013E121938 00013E122683	Device type i62 Talker i62 Talker i62 Messenger	Software version 2.2.22 2.2.22 2.2.17	Parameter version 13.16 13.16 13.16	Upgrade status	Online ✓	Latest number 54009 54004	

Figure 20: Ascom Device Manager Devices Tab

Select the **Numbers** tab as shown in **Figure 21**. Click on the **New** icon to add a new number **54008** in this example.

🗿 Belleville - As	scom WinPDM								3
File Device Nu	mber Template	e License Optio	ons Help						
Devices Number	s Templates L	icenses							
New Edit Delet	¢								
Device types:	Search for:		in: [Number	▼ Sho	ow all			
(All)	Number	Device type	Parameter v	Device ID	Online	Status	Saved	Last run tem	
i62 Messenger	54004	i62 Messenger	13.16	00013E122683		Synchronized	×		-
	54009	i62 Talker	13.16	00013E121938		Synchronized	~		
									Ŧ
									11

Figure 21: Ascom Device Manager Numbers Tab

There is a dialog box popping up as shown in **Figure 22.** Enter **54008** in the textbox of **Call number** parameter. Click **OK** to create the new number in the **Numbers** table.

Device type:	i62 Talker			
Parameter version:	13.16 👻			
Template:	None			
Prefix	:			
Single Call no	umber:	54008		
🔘 Range Start o	all number:			
Stop	all number:	[

Figure 22: Device Manager Add New Numbers

On the **Numbers** tab, the number **54008** is now shown up on the Number list as shown in **Figure 23**.

Devices Numbe	Templates	icenses						
New Edit Dele	Search for:		in:	Number	▼ Shc	wall		
(All)	Number	Device type	Parameter v	. Device ID	Online	Status	Saved	Last run tem
	54004	i62 Messenger	13.16	00013E122683		Synchronized	1	
62 Messenger		and the second se	13.16					
62 Messenger 62 Talker	54008	i62 Talker	10,10					
i62 Messenger i62 Talker	54008 54009	i62 Talker i62 Talker	13.16	00013E121938		Synchronized	~	

Figure 23: Device Manager with New Number Added

Right click on the newly created number **54008** and choose the **Associated Numbers** to associate the new number with the i62 physical device being inserted in **Figure 20**. Pop up **Associated Number** window will be as shown in **Figure 24**. Choose the i62 set to associate the number with and click **OK** to assign the number.

	vice to associ	ate with		0.1			
Device ID	Device t	Softwar	Parameter	Upgrad	Online	Latest	
00013E12	i62 Talker	2.2.22	13.16		1	54008	L,
00013E12	. i62 Talker	2.2.22	13.16			54009	

Figure 24: Associate a Number to Physical Set

Figure 25, below, shows the inserted i62 set with its assigned number 54008 in the Numbers table.

Devices Numbe	rs Templates Li	censes							
New Edit Dele	Kete Search for:		in:	Number	▼ Sh	ow all			
	Contraction of the second second second		107728	seren and a series of the seri	Contraction of the second				
(All)	Number	Device type	Parameter v	Device ID	Online	Status	Saved	Last run tem	
(All) 62 Messenger	Number 54004	Device type	Parameter v 13.16	Device ID 00013E122683	Online	Status Synchronized	Saved	Last run tem	
(All) i62 Messenger i62 Talker	Number 54004 54008	Device type i62 Messenger i62 Talker	Parameter v 13.16 13.16	Device ID 00013E122683 00013E1218E2	Online	Status Synchronized Synchronized	Saved ✓	Last run tem	
(All) i62 Messenger i62 Talker	Number 54004 54008 54009	Device type i62 Messenger i62 Talker i62 Talker	Parameter v 13.16 13.16 13.16	Device ID 00013E122683 00013E1218E2 00013E121938	Online	Status Synchronized Synchronized Synchronized	Saved	Last run tem	,

Figure 25: New Number with Associated i62 Set

Double click on the entry for the handset to be configured, select the **Network -> Network A**, an **Edit Parameters for 54008** Window will appear as shown in **Figure 26**. Fill in the parameters as highlighted in red.

Note: This setting is one of many ways to configure the network set up for the i62 handset. For more information how to configure this in a different way, refer to **Reference [2]**.



Figure 26: Network Parameters

Select the VoIP-> General menu, and enter the values highlighted in red as shown in the Figure 27. Click OK (not shown) to save the change.

evice type: i62 T	alker		
arameter version: 13.1	6		
Network	Name	Value	
Device	Replace Call Rejected with User Busy	Enable	
Brosonco	VoIP protocol	SIP	
Location	Codec configuration	G.711 A-law	(
VoID	Codec packetization time configuration	20	
A Conorol	Internal call number length	5	(
	Endpoint number	54008	
# SID	Endpoint ID	54008	(
Customization			
Headset			
Profiles			
A Normal			
Profile 1			
Profile 2			
Profile 3			
Profile 4			
Chanterte			

Figure 27: VoIP General Parameters

Select the VoIP->SIP menu point, and enter the values highlighted in red as shown in Figure 28. Click OK (not shown) to save the changes.

Device type: i62 Ta	alker		
Parameter version: 13.16	5		
Network	Name	Value	
Device	SIP proxy IP address	0.0.0.0	0
AUGIO	Secondary SIP proxy IP address	0.0.0.0	0
Presence	SIP proxy listening port	5070	6
Uocation	SIP proxy ID	sipl75.com:5070	6
VOIP	SIP proxy password	********	6
General	Send DTMF using RFC 2833 or SIP INFO	RFC2833	
H.323	Hold type	Inactive	
Customization	Registration identity	Endpoint number	
Headret	Authentication identity	Endpoint number	(
Profiles	Call forward locally	Enabled	
Shortcuts	MOH locally	Enabled	6
Janoi couta	Hold on Transfer	Disabled	
	Direct signaling	Disabled	(
	SIP Register Expiration	120	6

Figure 28: VoIP SIP Parameters

Note: For **SIP Register Expiration** parameter, it should be set at **120** as recommended by Ascom.

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7. Verification Steps

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

- Verify that the Ascom Wifi i62 telephone registers successfully with the CS 1000 SIP Line Gateway server and Call Server by using the CS 1000 Linux command line and CS 1000 Call Server overlay LD 32.
 - Log in to the SIP Line server as an administrator by using Avaya account.
 - Issue command "slgSetShowByUID [userID]" where userID is SIP Line user's ID being checked.
 [admin@sipl ~]\$ slgSetShowByUID 54008

```
=== VTRK ===
                          AuthId TN
UserID
                                                                               Clients Calls
SetHandle Pos ID SIPL Type
----- ---- ----- ----- ----- -----
_____ ____
5400854008104-00-00-01100x8fc4cf8SIP Lines
            StatusFlags = Registered Controlled KeyMapDwld SSD
              FeatureMask =
              CallProcStatus = 0
              Current Client = 0, Total Clients = 1
                == Client 0 ==
                IPv4:Port:Trans = 10.10.98.55:5060:udp
                Type = SIP3
               UserAgent = (Ascom i62/Ascom i62 2.2.22 \(2011-03-
30) release)
               x-nt-guid = 267d228547c1562399f1f743a2971fb5
RegDescrip =

      RegDescrip
      =

      RegStatus
      =

      PbxReason
      =

      OK
      SipCode

      SipCode
      =

      MTransc
      =

      (nil)
      Expire

      Expire
      =

      3600

      Nonce
      =

      f56a9946ba497bde7eb445efb518f4f1

      NonceCount
      =

      hTimer
      =

      0x8f64e60

      TimeRemain
      =

      Stale
      =

      MSec CLS
      = 0

      MSNV (MSEC-Never)

      Contact
      = sip:54008@10.10.98.55:5060

      KeyNum
      = 255

      AutoAnswer
      = NO

              Key Func Lamp Label
                     3 0 54008
126 0 2654008
9 0
29 0
              0
              1
              2
              3
```

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4	22	0	
5	2	0	54334
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	
== 5	Subscr	iption	Info ==
Subs	script	ion Eve	ent = None
Subs	script	ion Har	ndle = (nil)
Subs	scribe	Flag =	0

- Log in 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 104 0 0 1
IDLE REGISTERED 00
```

- Place a call from and to Ascom Wifi i62 telephone and verify that the call is established with 2-way speech path.
- During the call, use a pcap tool (ethereal/wireshark) at the SIP Line 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 the Section 2.1, with some exceptions outlined in Section 2.2. The Ascom Wifi i62 firmware version 2.2.22 is considered to be in compliance with Avaya CS 1000 SIP Line System Release 7.5.

9. Additional References

Product documentation for the Avaya CS 1000 products may be found at: <u>https://support.avaya.com/css/Products/</u>

Product documentation for the Ascom Wifi i62 products may be found at: <u>http://www.ascom.com</u>

[1] Avaya CS1000 Documents:

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 Sever 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] Ascom Wireless i62 Wifi Phone Documents:

Ascom i62 VoWifi Handset Quick Reference Guide Installation and Operation Manual, Portable Device Manager Configuration Manual, Ascom i62 VoWifi Handset

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