

Application Notes for Ascom Wifi i62 SIP Telephone firmware version 2.2.22 with Avaya Communication Server 1000 Release 7.5 – Issue 1.1

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 i62 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.

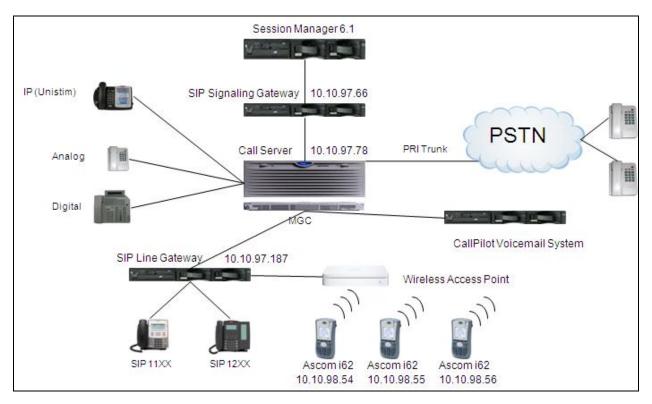


Figure 1: Network Configuration Diagram

4. Equipment and Software Validated

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 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)

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

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 <u>http://www.avaya.com</u>.

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.

avaya	Avaya Unified Communi	cations Manage	ment		Help Logo
Network Elements	Host Name: car2-sipl-ucm.bwdev.com	Software Version: 02.20)-SNAPSHOT(0000) User Na	ame admin	
- CS 1000 Services IPSec	Elements				
Patches SNMP Profiles Secure FTP Token	New elements are registered into the sec can optionally filter the list by entering a s		added as simple hyperlinks. (Click an element name to launch its m	anagement service. You
Software Deployment	Ę	Search Reset			
Administrative Users External Authentication	Add Edit Delete				· <u>22</u>
Password Security	Element Name	Element Type +	Release	Address	Description
Roles	1 EM on car2-cores	CS1000	7.5	*******	New element.
Policies Certificates	2 EM on car2-ssg-carrier	CS1000	7.5		New element.
Active Sessions Tools	3 EM on cpppm3	CS1000	7.5		New element.
Logs Data	4 car2-ssg-carrier.bwdev.com (member)	Linux Base	7.5		Base OS element.
	5 car2-sipl-ucm.bvwdev.com (primary)	Linux Base	7.5	*25.4725.40 3	Base OS element.
	6 🔲 car2-mas.bwdev.com (membe	<u>r)</u> Linux Base	7.5	**********	Base OS element.
	7 Car2-cores.bwdev.com (memb	er) Linux Base	7.5		Base OS element.
	8 car2-sps.bwdev.com (member) Linux Base	7.5	49345494370	Base OS element.
	9 cpppm3.bwdev.com (member)	Linux Base	7.5		Base OS element.
	im sinl75 hwwdav.com (member)	Linux Raea	7.5		Bass OQ
	Copyright 2002-2010 Avaya Inc. All rights res	anvad			

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 - Virtual Terminals - System - Alarms - Maintenance + Core Equipment - Peripheral Equipment - Peripheral Equipment - Peripheral Equipment - Peripheral Equipment - Pretwork - Interfaces - Engineered Values - Emergency Services - Goographic Redundancy + Software - Customers - Routes and Trunks - D-Channels - Digital Trunk interface - Digital and Numbering Plans - Electronic Switched Network - Flexible Code Restriction - Incoming Digit Translation - Properties - Keyorts - Lists - Froperties - Migration - Tools - Backup and Restore - Date and Time	CS1000 Element Manager Menaging: Control of the manne: admin System Overview System Overview IP Address: 10.10.97.78 Type: Awaya Communication Server 1000E CPPM Linux Version: 4121 Release: 750 Q +	Help Logout
+ 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 - Virtual Terminals - System + Alarms	Managing: <u>det 10.97.78</u> Username: admin <u>Oustomers</u> » Customer 00 » <u>Oustomer Details</u> » SIP Line Service SIP Line Service	
- Maintenance • Core Equipment - Peripheral Equipment • IP Network • Interfaces • Engineered Values • Emergency Services • Geographic Redundancy • Software • Security • Passwords • Policies • Login Options	Image: Signal Service User agent DN prefix 26 Optional features: Image: Nortel Multimedia	
	*Required Value Copyright © 2002-2011 Avaya Inc. All rights reserved.	Save Cancel

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.

avaya	CS1000 Element N	lanager				Help Logout
- UCM Network Services			s » New IP Telephony Node			
- Home			s » New IP Telephony Node			
- Links	New IP Telephony N	ode				
- Virtual Terminals	Step 1: Define the new Node	and its services.				
- System	You will also require p	re-configured serve	ers with appropriate application software	already deployed to host the	selected services.	
+ Alarms						
- Maintenance						
+ Core Equipment	Node ID:	510	* (0-9999)		A	
 Peripheral Equipment IP Network 	Node ID:	512	. (0-9999)			
- Nodes: Servers, Media Cards	Call server IP address:	10 10 97 78	* TLAN address type:	C ID-1 ant		
- Maintenance and Reports	Gan Server in address.	10.10.01.10	re/waddiess type.	-		
- Media Gateways				IPv4 and IPv6		
- Zones						
- Host and Route Tables	Embedded LAN (ELAN)		Telephony LAN (TLAN)			
 Network Address Translation (NA[*]) 						
- QoS Thresholds	Gateway IP address:	10.10.97.65	* Node IPv4 address:	10.10.97.187 *		
- Personal Directories					-	
- Unicode Name Directory + Interfaces	Subnet mask:	255.255.255.192	* Subnet mask:	255.255.255.192 *	-	
- Engineered Values						
+ Emergency Services			Node IPv6 address:			
+ Geographic Redundancy						
+ Software						
- Customers	Applications:	ISIP Line				
- Routes and Trunks		UNIStim Line Te	rminal Proxy Server (LTPS)			
- Routes and Trunks			teway (SIPGw, H323Gw)			
- D-Channels						
– Digital Trunk Interface		Personal Directo				
- Dialing and Numbering Plans - Electronic Switched Network		Presence Publis	sher		-	
- Flexible Code Restriction - Incoming Digit Translation	* Required Value.				Next > Cancel	
- Phones						
- Templates						
- Reports						
- Views						
- Lists						
- Properties						
- Migration 🔻						
< <u> </u>	Copyright © 2002-2011 Avaya Inc	. All rights reserved.				

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).

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Αναγα	c	CS1000 Element Manager	Help Logout
- UCM Network Services - Home - Links - Virtual Terminals - System + Alarms - Maintenance + Core Equipment	1	Managing: Managing:	-
 Peripheral Equipment IP Network 		Select to add Add Remove Make Leader Print Refresh	
- <u>Nodes: Servers, Media Cards</u> - Maintenance and Reports		Hostname ▲ Type Deployed Applications ELAN IP TLAN IPv4 TLAN IPv6 Role	
Personal Directories Unicode Name Directory Interfaces Engineered Values Seographic Redundancy Software Customers Routes and Trunks Poutes and Trunks Dochannels Digital Trunk Interface Dialing and Numbering Plans Electronic Switched Network Flexible Code Restriction Incoming Digit Translation Phones	н	Select from the list above and click Add to associate servers with this node. Selected servers must have identical application deployments.	
- Templates - Reports - Views - Lists - Properties - Migration	•	Copyright © 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 Ma	anager		Help Logout
- UCM Network Services - Home - Uriks - Virtual Terminals - System + Narms - Maintenance + Core Equipment - Perioheral Equipment	Node ID: 512 - SIP Line	me:admin 19 Telephony Nodes > Node Details > e Configuration Details Settings SIP Line Gateway Servic Line Gateway Application	<u>ê</u>	_
- IP Network	General		Virtual Trunk Network Health Monitor	
 <u>Nodes: Servers, Media Cards</u> Maintenance and Reports Media Gateways 	SIP domain name:	sipl75.com *	Monitor IP addresses (listed below) Information will be captured for the IP addresses listed	
– Zones – Host and Route Tables – Network Address Translation (NA	SLG endpoint name:	sipline	Monitor IP: Add	
 – QoS Thresholds ≡ – Personal Directories 	SLG Group ID:		Monitor addresses:	
- Unicode Name Directory + Interfaces - Engineered Values	SLG Local Sip port		Remove	
+ Emergency Services + Geographic Redundancy + Software	SIP Line Gateway Settings			
- Customers		Security policy: Secu	ity Disabled	
- Routes and Trunks			Ry Disabled	
- Routes and Trunks - D-Channels	Num	ber of byte re-negotiation: 0	÷	
- Digital Trunk Interface		Options: Clie	nt authentication 👻	
- Dialing and Numbering Plans - Electronic Switched Network - Flexible Code Restriction	* Required Value.		de on this page will NOT be Save Cancel	
- Incoming Digit Translation				
– Templates – Reports				
- Views - Lists				
- Lists - Properties				
- Migration				
4 III +	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 System » IP Network » IP Telephony Nodes » Node Details » SIP Line Configuration	
- Home		
– Links	Node ID: 512 - SIP Line Configuration Details	
- Virtual Terminals		
- System	General SIP Line Gateway Settings SIP Line Gateway Service	
+ Alarms	General Sir Line Gateway Setunds Sir Line Gateway Set Vice	
– Maintenance	SIP Line Gateway Service	
+ Core Equipment	Branch / GR 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: 0.0.0.0	
- Zones - Host and Route Tables	The IP address can have either IPv4 or IPv6 format based on the value of "TLAN	
 Host and Route Tables Network Address Translation (NA⁻ 	address type"	
- QoS Thresholds		
- Personal Directories	MO SLG IPv6 address:	
- Unicode Name Directory		
+ Interfaces	MO SLG port: 5070 (1 - 65535)	
- Engineered Values	MO SLG transport TCP 👻	
+ Emergency Services	WO SLG transport. TOP V	
+ 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	nde le adures type"	
- Routes and Trunks		
- Routes and Trunks	GR SLG IPv6 address:	
- D-Channels	5070	
- Digital Trunk Interface	GR SLG port: 5070 (1 - 65535) +	
- Dialing and Numbering Plans	* Required Value. Note: Changes made on this page will NOT be Save Cancel	
 Electronic Switched Network 	transmitted until the Node is also saved.	
 Flexible Code Restriction 		
 Incoming Digit Translation 		
- Phones		
– Templates		
- Reports		
- Views		
- Lists		
- Properties		
- Migration	Copyright © 2002-2011 Avaya Inc, All rights reserved.	
	oopyngin w 2002-2011 Araya III. Alliğins teset ved.	

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: 488 10.97.78 Username: admin System » IP Network » I <u>P Telephony Nodes</u> » Node Saved Node Saved	_
- System + Alarms - Maintenance + Core Equipment - Peripheral Equipment - IP Network - <u>Nodes: Servers, Media Cards</u> - Maintenance and Reports - Media Gateways	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.	
 - Zones - Zones - Host and Route Tables - Network Address Translation (NA[*] - QoS Thresholds - Personal Directories - Unicode Name Directory + Interfaces - Engineered Values 	Show Nodes You may initiate a transfer manually at a later time.	
Emergency Services Geographic Redundancy Software Customers Routes and Trunks III	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**.

AVAYA	cs	61000 Element Mana	iger			Help Logout
- UCM Network Services - Home - Links - Virtual Terminals - System + Aarms - Maintenance		Synchronize Configur	<u>IP Telephony Nodes</u> » Synchro ation Files (Node ID chronize their configuration	<512>)	This process transfers server INI f	files to selected
- Nonmertance - Core Equipment - Peripheral Equipment - IP Network - Nodes: Servers. Media Cards - Maintenance and Reports - Media Gateways - Zones - Host and Route Tables	II				Synchronization Status Sync required de to general LAN configurations, SNT	
Network Address Translation (N. - QoS Thresholds - Personal Directories - Unicode Name Directory Interfaces Engineered Values Emergency Services Geographic Redundancy Software	4	H323 Gatew ay settings, network c servers.	onnectivity related parameters	like ports and IP address, enab	ling or disabling services, or adding or	r removing application
- Customers - Routes and Trunks	-	✓ Copyright © 2002-2011 Avaya Inc. /	All 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	000 Element Manager	Help Logout			
UCM Network Services Home Links - Virtual Terminals System Aarms Maintenance Core Equipment Peripheral Equipment IP Network Nodes: Servers, Media Cards Maintenance and Reports Media Gateways Zones Host and Route Tables Network Address Translation (NA* CooS Thresholds Personal Directories Unicode Name Directory Interfaces	Managing 10.97.78 Username: admin Routes and Trunks » D-Channels D-Channels Maintenance D-Channel Diagnostics (LD 96) Network and Peripheral Equipment (LD 32, Virtual D-Channels) MSDL Diagnostics (LD 96) TMDI Diagnostics (LD 96) D-Channel Expansion Diagnostics (LD 48) Configuration Choose a D-Channel Number: 4 v and type: DCH v 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: RIse	5 Edit			
Customers Routes and Trunks Routes and Trunks D-Channels Digital Trunk Interface Dialing and Numbering Plans Electronic Switched Network Flexible Code Restriction Incoming Digit Translation Phones Temnlates	- Channel: 3 Type: DCH Card Type: DCIP Description: SIPI	Line Edit			

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.

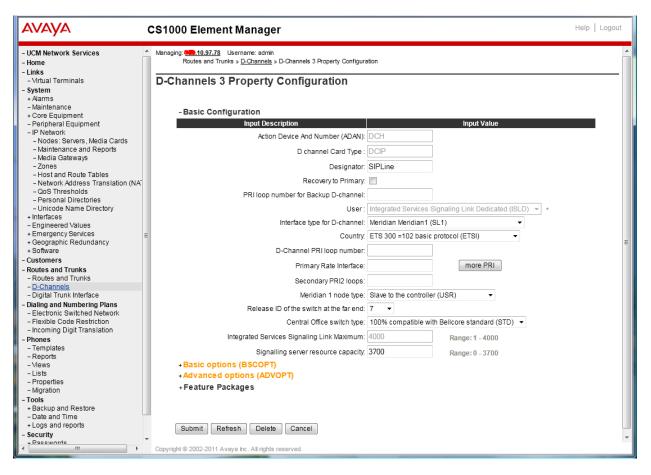


Figure 12: SIP Line D-Channel Property Configuration

- 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 (DV2O)	
– Links – Virtual Terminals	Diversion info. sent. rerouting requests processed (DV3I)	
- System	EuroISDN - div. info sent. rerouting req. processed (DV30)	
+ Alarms	Call transfer notification and invocation to EuroISDN (ECTO)	
– Maintenance + Core Equipment	Malicious call identification (MCID)	
– Peripheral Equipment	MCDN QSIG conversion (MQC)	
+ IP Network + Interfaces	Remote D-channel is on a MSDL card (MSL)	
– Engineered Values	Message waiting interworking with DMS-100 (MWI)	
+ Emergency Services + Geographic Redundancy	Network access data (NAC)	
+ Software	Network call trace supported (NCT)	
- Customers - Routes and Trunks	Network name display method 1 (ND1)	
- Routes and Trunks	Network name display method 2 (ND2)	
– <u>D-Channels</u> – Digital Trunk Interface	Network name display method 3 (ND3)	
- Digital Hunk Interface	Name display - integer ID coding (NDI)	
- Electronic Switched Network		
 Flexible Code Restriction Incoming Digit Translation 	Name display - object ID coding (NDO)	
- Phones	Path replacement uses integer values (PRI)	
– Templates – Reports	Path replacement uses object identifier (PRO)	
- Views	Release Link Trunks over IP (RLTI)	
– Lists – Properties	Remote virtual queuing (RVQ)	
- Migration	Trunk anti-tromboning operation (TAT) 📃	
- Tools	User to user service 1 (UUS1)	
+ Backup and Restore - Date and Time	NI-2 name display option. (NDS) 📃	
+ Logs and reports	Message waiting indication using integer values (QMWI) 📃	
- Security + Passwords	Message waiting indication using object identifier (QMWO) 📃	
+ Policies + Login Options	User to user signalling (UUI) 📃	
	Return - Remote Capabilities Cancel	×
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	😜 Intern	et 🔍 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://cpppm3.bvwd	.com ′em Web_6-0/SEC 🔎 - 😮 Certificate er 🗟 Ĉ × 🛛 🧀 Element Manager 🛛 ×
Αναγα	CS1000 Element Manager Help Logout
- UCM Network Services - Home - Links - Virtual Terminals - System - Alarms - Maintenance + Core Equipment - Peripheral Equipment - IP Network - Nodes: Servers, Media Cards - Maintenance and Reports - Media Gateways - Zones - Host and Route Tables - Network Address Translation (N/, - QoS Thresholds - Personal Directories - Unicede Name Directory	
- Interfaces - Application Module Link - Value Added Server - Property Management System - Engineered Values	* Required value.

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.

C https://cpppm3.bvwdev	com/emWeb_6-0/S 🔎 – 😮 Certificate e 🗟 🖒 🗙 🤗 Element Manager 🛛 🖌 🎯
Αναγα	CS1000 Element Manager Help Logout
– Virtual Terminals – System + Alarms – Maintenance	Managing: <u>wef.10.97.78</u> Username: admin System » Interfaces » <u>Value Added Server</u> » <u>Add Value Added Server</u> » Ethernet Link
- Nameriance - Core Equipment - Peripheral Equipment - IP Network - Nodes: Servers, Media Cards - Maintenance and Reports - Media Gateways - Zones - Host and Route Tables - Network Address Translation (NA* - QoS Thresholds - Personal Directories - Unicode Name Directory - Interfaces - Application Module Link - <u>Value Added Server</u> - Property Management System - Engineered Yalues	Ethernet Link Value added server ID: 33 • (16 - 127) Ethernet LAN Link: 33 • ELAN port configured in ADAN Application security: Interval: 1 • Time interval for checking the link for overload in five second increments Message count threshold: 9999 • (10 - 9999)
+ Geographic Redundancy + Software	* Required value. Save Cancel
- Customers	Copyright © 2002-2011 Avaya Inc. All rights reserved.

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	Help Logout
- UCM Network Services - Home - Links - Virtual Terminals - System - Alarms - Maintenance	Managing: <u>166.10.97.78</u> Username: admin System » IP Network » <u>Zones</u> » <u>Bandwidth Zones</u> » Zone Basic Property and Bandwidth Management Zone Basic Property and Bandwidth Management Input Description Input Value	
Core Equipment Core Equipment Peripheral Equipment IP Network Nodes: Servers, Media Cards Maintenance and Reports Media Gateways Zones Host and Route Tables Network Address Translation (NA QoS Thresholds Personal Directories Unicode Name Directory Interfaces Application Module Link Value Added Sever Property Management System	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) •	E
- Engineered Values - Engineered Values - Engency Services - Geographic Redundancy - Software III	* Required value. Save Copyright © 2002-2011 Avaya Inc. All rights reserved.	Cancel

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.

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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 config	guration
Cher the Submit Outlos		udding the route	and buve coming	5 ^{arano} 11

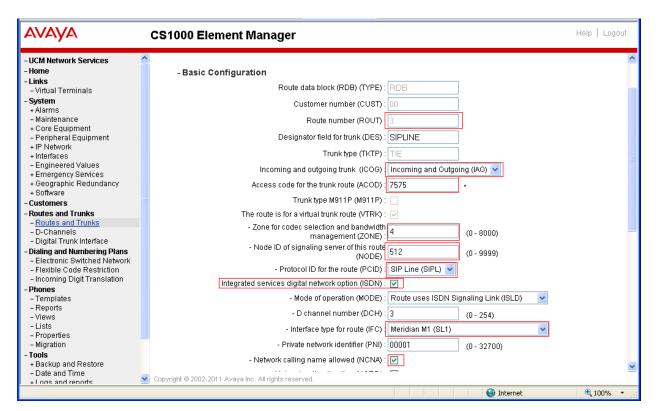


Figure 17: SIP Line Route Configuration

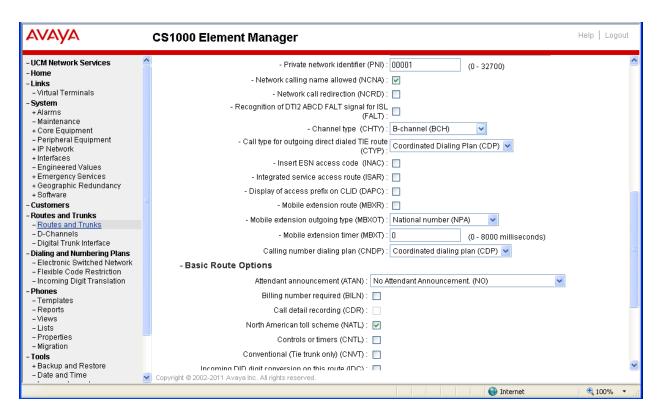


Figure 18: SIP Line Route Configuration (cont)

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, 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.

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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.

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

Click on the Save button to complete adding virtual trunks for 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 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_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. 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 INI	imber Template	License Options	Help					
Devices Numbe	rs Templates Lice	nses						
DX P								
Delete Upgrade	software Cancel							
beiete opgibae								
Device types:	Search for:		in: Device I	ID 👻 Sh	ow all			
No. of Street, Str	Device ID	Device type	Software version	Parameter version	Upgrade status	Online	Latest number	
i62 Messenger	Device ID 00013E1218E2	Device type i62 Talker	Software version	Parameter version 13.16	Upgrade status	Online	Latest number	-
i62 Messenger					Upgrade status	Online	Latest number	
(All) i62 Messenger i62 Talker	00013E1218E2	i62 Talker	2.2.22	13.16	Upgrade status	Online ✓		
i62 Messenger	00013E1218E2 00013E121938	i62 Talker i62 Talker	2.2.22 2.2.22	13.16 13.16	Upgrade status	Online ✓	54009	

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 - A	scom WinPDM								×
File Device Nu	umber Template	e License Optic	ons Help						
Devices Numbe	rs Templates L	icenses							
New Edit Dele	K te								
Device types:	Search for:		in:	Number	• S	how all			
(All)	Number	Device type	Parameter v	. Device ID	Online	Status	Saved	Last run tem	
i62 Messenger i62 Talker	54004	i62 Messenger	13.16	00013E122683		Synchronized	1		
	54009	i62 Talker	13.16	00013E121938		Synchronized	~		
									Ŧ
1									1

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.

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Device type:	i62 Talker	
Parameter version:	13.16	-
Template:	None	•
Prefi	x:	
Single Call r	number:	54008
🔘 Range Start	call number:	
Stop	call 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**.

New Edit Delete Device types: Search for: in: Number Show all 62 Messenger 0404 i62 Messenger 13.16 00013E122683 Synchronized Value 62 Talker 54004 i62 Talker 13.16 00013E122683 Synchronized Value	Devices Numbe	ers Templates L	icenses						
62 Messenger 54004 i62 Messenger 13.16 00013E122683 Synchronized ✓ 62 Talker 54008 i62 Talker 13.16 00013E122683 Synchronized ✓	Device types:	Search for:	2.4.1.4	1	- Keiner auf der Kein			5	
62 Talker 54008 i62 Talker 13.16		Number				Online			Last run tem
54009 i62 Talker 13, 16 000 13E12 1938 Synchronized	62 Messenger	54004		10,10	000136122003		Syrici i Offized	v	
	101 101 000 00 000 Die 1013	2 C 2 C 2 C	Contraction of the second second second	13.16					

Figure 23: Device Manager with New Number Added

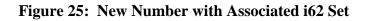
Right click on the newly created number **54008** and choose the **Associated Numbers** to associate the new number with the i62 physical device being inserted in **Figure 20**. Pop up **Associated Number** window will be as shown in **Figure 24**. Choose the i62 set to associate the number with and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

Device ID	Device t	Softwar	Parameter	Upgrad	Online	Latest	
	i62 Talker		13.16		\checkmark	54008	i L
00013E12	i62 Talker	2.2.22	13.16			54009	
			in: Device	• TD	▼ SI	now all	
Search for:			III. DEVICE	10		10 VV CIII	

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 L	icenses							
New Edit Dele	Search for:		in:	Number	▼] She	ow all			
(All)	Number	Device type	Parameter v	. Device ID	Online	Status	Saved	Last run tem	
52 Messenger	54004	i62 Messenger	13.16	00013E122683		Synchronized	1		
52 Talker	54008	i62 Talker	13.16	00013E1218E2		Synchronized	1		
	54009	i62 Talker	13.16	00013E121938		Synchronized	~		



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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]**.

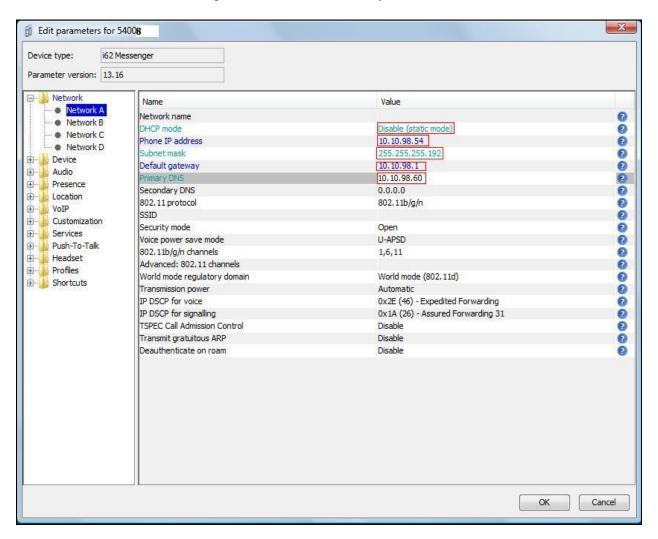


Figure 26: Network Parameters

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.

Device type:	i62 Talker			
arameter version: 13.16				
0 Network	Name		Value	
B- Device B- Audio	Replace Call Rejected with	th User Busy	Enable	
Presence	VoEP protocol		SIP	
Location	Codec configuration		G.711 A-law	
VoIP	Codec packetization time		5	
General	Internal call number leng	th		
- • H.323	Endpoint number		54008	
. SIP	Endpoint ID		54008	
L Customizatio	n			
Headset				
Profiles				
Normal				

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	Talker		
Parameter version: 13	16		
Network	Name	Value	
Device Audio Audio Presence Location VoIP General H.323 Customization Headset Profiles Shortcuts	SIP proxy IP address	0.0.0.0	0
	Secondary SIP proxy IP address	0.0.0.0	0
	SIP proxy listening port	5070	0
	SIP proxy ID	sipl75.com:5070	0
	SIP proxy password	****	0
	Send DTMF using RFC 2833 or SIP INFO	RFC2833	6
	Hold type	Inactive	0
	Registration identity	Endpoint number	
	Authentication identity	Endpoint number	0
	Call forward locally	Enabled	0
	MOH locally	Enabled	0
J Shortcuts	Hold on Transfer	Disabled	0
	Direct signaling	Disabled	6
	SIP Register Expiration	120	0

Figure 28: VoIP SIP Parameters

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

DNS entry is required to resolve the domain (sipl75.com) into IP address (10.10.97.187) as required. In this example configuration, the DNS entry is shown in figure bellow.

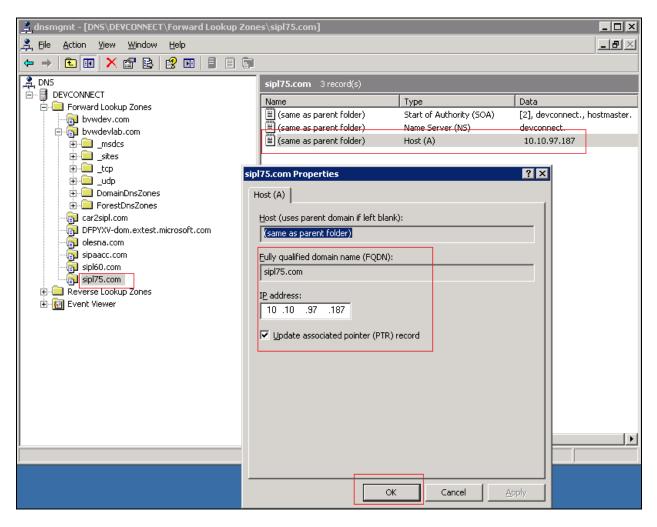


Figure 29: DNS resolution for sipl75.com domain

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
----- ---- ----- ----- ------ -----
----- ------
             54008 54008 104-00-00-01 1 0
0x8fc4cf8 SIP 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
      =

      Pbxreason
      =

      Nonce
      =

      (nil)
      =

      Expire
      =

      3600
      =

      Nonce
      =

      f56a9946ba497bde7eb445efb518f4f1

      NonceCount
      =

      1
      =

      0
      =

      0
      =

             ClientGUID = 0

MSec CLS = MSNV (MSEC-Never)

Contact = sip:54008@10.10.98.55:5060

KeyNum = 255

AutoAnswer = NO
           Key Func Lamp Label
            0 3 0 54008
1 126 0 2654008
            2 9 0
```

4 22 0 5 2 0 54334 17 16 0 18 18 0 19 27 0 20 19 0 21 52 0 24 11 0 25 30 0 26 31 0
17 16 0 18 18 0 19 27 0 20 19 0 21 52 0 22 25 0 24 11 0 25 30 0
18 18 0 19 27 0 20 19 0 21 52 0 22 25 0 24 11 0 25 30 0
19 27 0 20 19 0 21 52 0 22 25 0 24 11 0 25 30 0
20 19 0 21 52 0 22 25 0 24 11 0 25 30 0
21 52 0 22 25 0 24 11 0 25 30 0
22 25 0 24 11 0 25 30 0
24 11 0 25 30 0
25 30 0
26 31 0
== Subscription Info ==
Subscription Event = None
Subscription Handle = (nil)
SubscribeFlag = 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

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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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