Test Report of Certification

SpectraLink 8440 VoWiFi Handset

with

OpenScape Voice V5.00.01.ALL.11_PS0017.E08

Test Status: review partner
Release Date: 2011-09-XX

Issued by:
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Contents:

History of Change

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1 Overview

1.1 Test Object

1.1.1 Basic Equipment

Test system: OpenScape Voice V5 (formerly called HiPath 8000)

Software Version: OpenScape Voice V5.00.01.ALL.11_PS0017.E08

Gateways RG8702: V1.3 13.31.02.25

Wireless Controller	C2400: 07.41.05

1.1.2 Polycom VoWiFi

Certification:	Test of interface functionality between the OpenScape Voice and the Polycom SpectraLink VoWiFi handset
Test Equipment:	OpenScape Voice in combination with an RG8702 (PRI) gateway and the Siemens HiPath Wireless C2400 Controller and Access Points (AP)
Software Release:	8440: V 4.0.0.20561
HW / FW Release:	
Manufacturer:	Polycom
Description:	The 8440 VoWiFi handset functions as a SIP device registered on the Openscape Voice Server.
Documentation:	- Configuration Manual: Polycom SpectraLink
Test Network:	Test network of HiPath Ready Lab Brussels
Test Configuration:	See section 2.3

1.2 Test Strategy

This certification test for the **Polycom** phones listed below with the **Siemens OpenScape Voice V5** focused on the verification of the SIP interface in the following scenarios:

- Basic phone configuration and registration
- Basic calls
- Telephony feature verification
- Audio features, including codecs and DTMF
- Basic WLAN test were performed

Regarding the compliance with acoustic standards no tests were performed.

1.2.1 Test Intensity

Scopes of the tests are to execute / to verify the solution performs within the limits of the system requirements, targeting the end product. To accomplish this, feature and solution based test cases are created, inspected, and executed under a real system environment (mirroring as close as possible real customer's environment).

Note:
The testing of the product with regard to compliance to requirements for Product Safety, EMV, Network Access Interfaces and Radiation Protection were not performed.
Siemens Enterprise Communications therefore assumes no responsibility for the compliance to these requirements.

1.2.2 Measuring / Test Instruments

Tracing and monitoring available on all IP phones, servers, gateways and laptops. No special hardware required.

1.3 Realisation Data

Test Preparation: September 2011

Test Duration: september 14th – 20th, 2011

Test Location: Siemens Enterprise Communications
Demeurslaan 134
1654 Huizingen
International Solution Lab

Test Personnel: Siemens OpenScape Voice:
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1.4 Test Results Summary

For details please have a look at the test results.

1.4.1 Problems

1) Occasionally, when the DUT is put on hold, the MOH heard on the DUT "chopped". Traces were made on the Media Server, the MOH streams is ok there.

1.4.2 Restrictions

1)

2)

1.4.3 Remarks

1) The SpectraLink devices understand DNS-SRV-records. Devices have been configured with SRV-record for OSV and RG8700. Automatic registration on RG8700 in case of OSV failure was successfully tested.

2) Devices have been tested with TLS activated. Please use following configuration guide to activate TLS.

TLS Spectralink OSV5

3) SRTP between Polycom and Siemens devices is not supported due to different standards used (SDES vs MIKEY). SRTP between Polycom devices however is possible and has successfully been tested. For more information on activating srtp, please refer to the Spectralink admin manual.

2 Configuration

2.1 Polycom devices

The Polycom devices have been configured according to the manual.

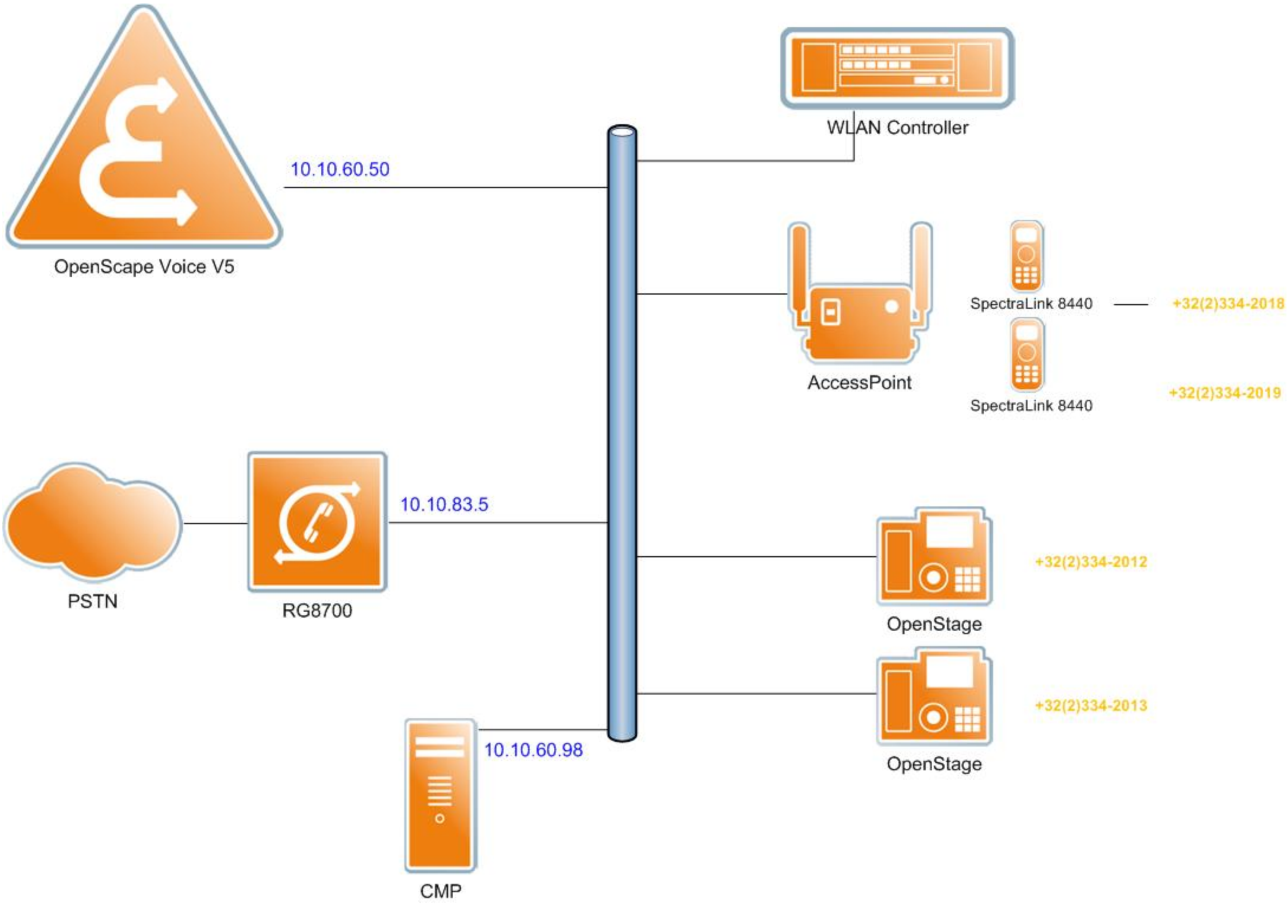
2.2 OpenScape Voice

- HW Version: Fujitsu Siemens RX330
- SW Version: V5.00.01.ALL.11_PS0017.E08
- IP phones
  - OpenStage 20/40/60/80 V1 R4.19.0
- RG8702 v1.3 13.31.02.25

2.3 HiPath Wireless Controller

- HW Version : C2400
- SW Version: V 7.41.05.003
- Access Points: AP3610

2.4 Configuration Block Diagram



Ax = +3223342018/2019 = Polycom SpectraLink handsets = DUT (device under test)
Oy = +3223342012/2013 = OpenStage IP phones
Eu = external PSTN phone u

x, y, z, u are digits from 0 to 9

3 Test Results in Detail

3.1 Tests

The syntax of the abbreviations used in the test cases :

Ax = +3223342018/2019 = Polycom SpectraLink handsets = DUT (device under test)
Oy = +3223342012/2013 = OpenStage IP phones
Eu = external PSTN phone u

x, y, z, u are digits from 0 to 9

3.1.1 Connectivity and Basic Operation

Test Case	Test Description	Result	Comment
1	Power up the handset and verify that the phone obtains a valid IP address from the **DHCP** server.	OK	
2	Connect a PC to the lab LAN and verify that access to the GUI of the test phone is possible.	OK	
3	Program the phone via GUI with the OSV **registrar** information and verify that the phone registers	OK	
4	Change the OSV subscriber settings so that **Digest Authentication** is required for the registration. Verify that the phone does not register.	OK	
5	Add the information for **HTTP Digest Authentication** to the test phone settings and verify that the phone registers	OK	
6	Verify that the test phone displays the local date and time correctly that is provided by the lab's SNTP server (10.10.1.12).	OK	
7	The first node of the OpenScape Voice server is put out of service, which means that on the second node the backup registrar IP address is coming up.	OK	
8	The first node of the OpenScape Voice server is put in service again, which means that on the first node the registrar IP address is coming up.	OK	
9	The DUT is **registered** on the same number as the O1 phone. This can be used to use the two phones in parallel (like is done sometimes with a hardphone and a soft client).	OK	

3.1.2 Basic call

For every test the HTTP Digest Authentication was enabled on the IP phones.

Test Case	Test Description	Result	Comment
10	Initiate a call from the DUT to **internal** subscriber A2. Verify that A2 is ringing (DUT receives ring back) and that the displays on the DUT and A2 show the correct called/calling number/name information.	OK	CLIP and calling name OK on both phones.
11	From the previous test case answer the call at A2 and verify speech path between both phones. Verify that the phone displays show the correct information after the call connected.	OK	
12	From the previous test case **disconnect** the call at the DUT and verify that both phones return to idle state.	OK	
13	Repeat the previous call, but disconnect the DUT before A2 answers. Verify that the DUT returns to idle state.	OK	A2 shows missed call
14	Initiate a call from A2 to the DUT. Verify that the DUT is ringing (A2 receives ring back) and that the displays on the DUT and A2 show the correct called/calling number/name information.	OK	CLIP and calling name OK on both phones.
15	From the previous test case answer the call at the DUT and verify speech path between both phones. Verify that the phone displays show the correct information after the call connected.	OK	
16	From the previous test case disconnect the call at the DUT and verify that both phones return to idle state.	OK	
17	Initiate a call from the DUT to **internal** subscriber O1. Verify that O1 is ringing (DUT receives ring back) and that the displays on the DUT and O1 show the correct called/calling number/name information.	OK	CLIP and calling name OK on both phones.
18	From the previous test case answer the call at O1 and verify speech path between both phones. Verify that the phone displays show the correct information after the call connected.	OK	
19	From the previous test case **disconnect** the call at the DUT and verify that both phones return to idle state.	OK	
20	Repeat the previous call, but disconnect the DUT before O1 answers. Verify that the DUT returns to idle state.	OK	DUT shows missed call
21	Initiate a call from O1 to the DUT. Verify that the DUT is ringing (O1 receives ring back) and that the displays on the DUT and O1 show the correct called/calling number/name information.	OK	CLIP and calling name OK on both phones.

22	From the previous test case answer the call at the DUT and verify speech path between both phones. Verify that the phone displays show the correct information after the call connected.	OK	
23	From the previous test case disconnect the call at the DUT and verify that both phones return to idle state.	OK	
24	Initiate a call from the DUT to an **external number**. Verify that the external phone is ringing (DUT receives ring back) and that the displays on the DUT and the external phone show the correct called/calling number.	OK	
25	From the previous test case answer the call at the external phone and verify speech path between both phones. Verify that the phone displays show the correct information after the call connected.	OK	
26	Initiate a call from an **external number** to the DUT. Verify that the DUT is ringing (external phone receives ring back) and that the displays on the DUT and the external phone show the correct called/calling number.	OK	
27	From the previous test case answer the call at the DUT and verify speech path between both phones. Verify that the phone displays show the correct information after the call connected.	OK	

3.1.3 Telephony features

Test Case	Test Description	Result	Comment
28	Initiate a call from the DUT to internal subscriber A2. Answer the call at A2. Put the DUT **on hold** and verify that it receives Music-on-hold.	OK	
29	From the previous test case retrieve the DUT from hold and verify speech path between the DUT and A2.	OK	
30	Initiate a call from internal subscriber A2 to the DUT. Answer the call at the DUT. From the DUT put A2 **on hold** and verify that it receives Music-on-hold.	OK	
31	Initiate a call from the DUT to internal subscriber A2. Answer the call at A2. Put the DUT **on hold** and verify that it receives Music-on-hold.	OK	
32	From the previous test case retrieve the DUT from hold and verify speech path between the DUT and A2.	OK	

33	Initiate a call from internal subscriber A2 to the DUT. Answer the call at the DUT. From the A2 put DUT **on hold** and verify that A2 receives Music-on-hold.	NOK	MOH is “chopped”
34	From the previous test case retrieve DUT from hold and verify speech path between the DUT and A2.	OK	
35	Initiate a call from the DUT to internal subscriber O1. Answer the call at O1. Put the DUT **on hold** and verify that it receives Music-on-hold.	OK	
36	From the previous test case retrieve the DUT from hold and verify speech path between the DUT and O1.	OK	
37	Initiate a call from internal subscriber O1 to the DUT. Answer the call at the DUT. From the DUT put O1 **on hold** and verify that it receives Music-on-hold.	OK	
38	From the previous test case retrieve the DUT from hold and verify speech path between the DUT and O1.	OK	
39	Initiate a call from internal subscriber O1 to the DUT. Answer the call at the DUT. From the DUT put O1 **on hold** and verify that it receives Music-on-hold.	OK	
40	From the previous test case retrieve the DUT from hold and verify speech path between the DUT and O1.	OK	
41	Initiate a call from the DUT to internal subscriber A2. Answer the call and initiate **consultation** at A2. Verify that the DUT receives Music-on-hold.	OK	
42	From the previous test case return from consultation and verify speech path between the DUT and A2.	OK	
43	Initiate a call from internal subscriber A2 to the DUT. Answer the call and initiate **consultation** at the DUT. Verify that A2 receives Music-on-hold while the DUT receives dial tone.	NOK	MOH is “chopped”
44	From the previous test case dial O1 at the DUT. Answer the call at O1. Verify that the DUT can **toggle** between A2 and O1.	OK	
45	From the previous test case initiate a **supervised transfer** at the DUT so that A2 and O1 are connected. Verify that A2 and O1 have speech path, the displays are correct, and that the DUT returns to idle state.	OK	
46	Initiate a **supervised transfer** at the O1 so that the DUT and O2 are connected. Verify that the DUT and O2 have speech path, the displays are correct, and that the DUT returns to idle state.	OK	

47	From the previous test case dial O1 at the DUT. Answer the call at O1. Verify that the DUT can **toggle** between O1 and O2.	OK	
48	From the previous test case initiate a **supervised transfer** at the DUT so that O1 and O2 are connected. Verify that O1 and O2 have speech path, the displays are correct, and that the DUT returns to idle state.	OK	
49	Initiate a call from the DUT to internal subscriber A1. Answer the call and initiate consultation at the DUT. Dial O1 and perform a **blind transfer** from A1 to O1. Answer O1 and verify that A1 and O1 have speech path, the displays are correct, and that the DUT returns to idle state.	OK	
50	Initiate a call from the O1 to the DUT. Answer the call on the DUT. Perform a **blind transfer** from the DUT to O2. Answer O2 and verify that O1 and O2 have a speech path, the displays are correct, and that the DUT returns to idle state.	OK	
51	Initiate a call from the DUT to the O2. Answer the call on O2. Perform a **blind transfer** from O2 to O3. Answer on O3 and verify that O3 and the DUT have a speech path, the displays are correct, and that the O2 returns to idle state	OK	
52	Initiate a call from the internal subscriber O1 to the O2. Answer the call on O2. Perform a **blind transfer** from O2 to the DUT. Answer on the DUT and verify that O1 and the DUT have a speech path, the displays are correct, and that the O2 returns to idle state	OK	
53	From the previous test case invoke the **last number redial** function on the DUT and verify that it calls O1.	OK	
54	Initiate a call to the DUT from an **external** number. Answer the call, then disconnect. Verify that the external number can be called from the call history list.	OK	
55	Initiate a call from the DUT to the internal subscriber O1. Answer the call and initiate a **three-way conference** from the DUT (conference master) with A1. Verify that all parties have speech path and that the displays on the phones indicate the conference.	OK	
56	From the previous test case **release the conference master** (= DUT). Verify that the O1 and A1 are in two-party talk and the displays are updated accordingly.	OK	
57	Initiate a call from the O1 to the internal subscriber DUT. Answer the call and initiate a **three-way conference** from the O1 (conference master) with A1. Verify that all parties have speech path and that the displays on the phones indicate the conference.	OK	Only master shows conference, other parties show call to master

58	From the previous test case **release the conference master** (= O1). Verify that the DUT and A1 are in two-party talk and the displays are updated accordingly.	OK	
59	Call the O1 from the DUT after the **Do-Not-Disturb** function was activated on O1. Verify that the call is rejected (phone based DND).	OK	No visual feedback on DUT, only auditory.
60	Call the DUT from O1 after the **Do-Not-Disturb** function was activated via a service code. Verify that the call is rejected (system based DND).	OK	
61	Activate **call forwarding (CFU)** on the OpenScape Voice server to A1. Call the DUT from O1 and verify that the call is forwarded to A1 (server based forwarding).	OK	
62	Activate **call forwarding (CFNR)** on the OpenScape Voice server to A1. Call the DUT from O1 and verify that the call is forwarded to A1 (server based forwarding).	OK	
63	Activate **call forwarding (CFB)** on the OpenScape Voice server to A1. Call the DUT from O1 and verify that the call is forwarded to A1 (server based forwarding).	OK	
64	Activate **call forwarding (CFU)** on the DUT to A1. Call the DUT from O1 and verify that the call is forwarded to A1 (device based forwarding).	OK	
65	Activate **call forwarding (CFNR)** on the DUT to A1. Call the DUT from O1 and verify that the call is forwarded to A1 (device based forwarding).	OK	
66	Activate **call forwarding (CFB)** on the DUT to A1. Call the DUT from O1 and verify that the call is forwarded to A1 (device based forwarding).	OK	
67	Put the DUT and O1 in the same pickup group. Call O1 from A2. While O1 is ringing, dial the **Group Pick-up** code (*7) from the DUT and verify that speech path to A2 is established and the display shows correct caller information.	OK	
68	Call the DUT from O1. While connected, call the DUT from A1 and verify that a **call waiting** indication is presented on the DUT that shows the calling party information.	OK	
69	From the previous test case accept the waiting call and verify that speech path is established between the DUT and A1. Verify that O1 is put on hold.	OK	
70	O1 has **call waiting** disabled. O1 is on the call with O2 and the DUT tries to call O1.	OK	User hears busy tone on the DUT.
71	Call O1 from the DUT and **reject** the call at O1. Verify that the DUT indicates the call rejection.	OK	User hears short bust tone and call ends.
72	Call O1 from the DUT and **deflect** the call to O2. Verify that the DUT indicates the call deflection.	OK	Display info on DUT gets update when the call is deflected.

73	Make the DUT busy and then call it from A2. Verify that the call is forwarded to the voicemail system (Xpressions) and that the **message waiting indication** (**MWI**) on the DUT is turned on.	OK	
74	From the previous test case retrieve the voicemail message and verify that the MWI is turned off.	OK	
75	While the MWI is lit on the DUT, disconnect the DUT from power and force a reboot. Verify that after the reboot is complete, the MWI is turned on.	OK	
76	While the MWI is lit on the DUT, reboot the Xpressions server. Verify that after the reboot is complete, the MWI is turned on.	OK	
77	The O1 subscriber does call the DUT. The DUT does not answer and the O1 comes into the voice mailbox of the DUT. The O1 subscriber leaves a voice message. The DUT receives a MWI. The DUT calls the call-back number of XPR and reads its message. After reading and deleting its message the MWI is turned off.	OK	
78	The DUT is put in an **MLHG** (multi-line hunt group) together with O1, and A2.	OK	
79	Large conference call between O1, O2, O3, A1, A2 and A2 (more than three parties involved). The conference initiator is O1.	OK	
80	O1 is busy. DUT calls O1 and activates **CCBS**.	OK	
81	DUT calls O1. O1 does not respond. DUT activates **CCNR**.	OK	
82	DUT is busy. O1 calls DUT and activates **CCBS**.	OK	
83	O1 calls DUT. DUT does not respond. O1 activates **CCNR**.	OK	
84	DUT parks a call and retrieves a call.	OK	
86	External party calls O1, O1 does blind transfer to DUT	OK	
87	External party calls O1, O1 does semi-attended transfer to DUT	OK	
88	External party calls O1, O1 does attended transfer to DUT	OK	
89	DUT has simultaneous ringing activated with O2. O1 calls DUT, both DUT and O2 ring.	OK	

90	O2 has simultaneous ringing activated with DUT. O1 calls O2, both DUT and O2 ring.	OK	

3.1.4 Audio features

Test Case	Test Description	Result	Comment
91	Configure A2 to use the G.729A codec only. Call the DUT from A2 and verify that the connection is established with G.729A (use Wireshark).	OK	
92	Configure the DUT for DTMF transmission via RFC 2833. Verify that from and to the DUT DMTF “telephony events” are sent.	OK	
93	Configure the DUT for DTMF transmission via RFC 2833. Verify that the Xpressions voicemail system can be accessed via DTMF.	OK	
94	Configure the DUT for DTMF transmission via RFC 2833. Verify that the DTMF tones are sent to and received from the PSTN.	OK	Remark: gateway supports RFC 2833

3.2 Remarks

Meanings of Abbreviations:

OK	Test case successful
NOK	Test case NOT successful
NA	Test case not applicable
NP	Test case not processed
NS	Situation not supplied
N *X	Error / restriction with description
* X	Remark to Functionality
DUT	Device Under Test
CFU	Call Forwarding Unconditional
CFNR	Call Forwarding on No Reply
CFB	Call Forwarding on Busy
MLHG	Multi Line Hunt Group
MoH	music-on-hold
DND	Do Not Disturb
AP	Access Points

4 Configuration Data

4.1 OpenScape Voice

4.1.1 System Basics

Configuration of the OpenScape Voice.

4.2 Polycom VoWiFi

4.2.1 Documentation

Polycom Documentation is downloadable from the Polycom Support website.

4.2.2 Basic Configuration

Handsets were configured as described in the manual.

4.3 Wireless network

Wireless network settings:

5 Confirmation

Testing personnel confirms that all the test cases were performed and that the results were as described in this document.

	Karel Eeckelaert
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