SIEMENS

Enterprise Networks

Test Report of Certification

POLYCOM

Polycom spectralink 8440 VoWiFi handset

with

SIEMENS

HiPath 4000 V6.0

Test Status: Released
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Issued by:
Siemens Enterprise Communications GmbH & Co.KG
Hofmannstraße 51
D-81359 Munich

Owner:	Eddy De Braekeleer
Department:	SEN Service PS
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History of Change

Date	***Description***	***Name***
August 2011	Initial Version	Eddy De Braekeleer SEN Service PS E-Mail: eddy.debraekeleer@siemens-enterprise.com Phone: +32.2.406.7316
August 25rd 2011	Review	Eddy Sterckx E-Mail: eddy.sterckx@siemens-entreprice.com Phone: +32.2.406 7197
August 25rd – September 2th, 2011 October 10th, 2011 October 20th, 2011 November 17th, 2011	Certification tests	Eddy Sterckx E-Mail: eddy.sterckx@siemens-entreprice.com Phone: +32.2.406 7197 Daniel Van Riet E-Mail: daniel.vanriet@siemens-entreprice.com Phone: +32 2 406 7336 Michel Lambrecht E-Mail: michel.lambrecht@siemens-enterprise.com phone: + 32 2 406 7325
November 18th, 2011	Review	Eddy Sterckx E-Mail: eddy.sterckx@siemens-entreprice.com Phone: +32.2.406 7197 Polycom: Oliver Wick Email: oliver.wick@Polycom.com Phone:
November 21, 2011	Final review	Eddy De Braekeleer SEN Service PS E-Mail: eddy.debraekeleer@siemens-enterprise.com Phone: +32.2.406.7316
November 22, 2011	Release	Eddy De Braekeleer SEN Service PS E-Mail: eddy.debraekeleer@siemens-enterprise.com Phone: +32.2.406.7316

1 Overview

1.1 Test Object

1.1.1 Basic Equipment

Test system:	Hipath 4000 CPCI
Software Version:	RMX V6 R1.10.0
Gateways	STMI: L0-T3R.62.001-007 LW:pzksti40 38.001-007
Wireless Controller	C2400: 07.41.02.0009 AP3610

1.1.2 Polycom Spectralink VoWiFi

Certification:	Test of interface functionality between the Hipath 4000 and the 8440 VoWiFi handset
Test Equipment:	Hipath 4000 in combination with an Siemens HiPath Wireless C2400 Controller and Access Points (AP)
Software Release:	8440: V 4.0.0.20561
HW / FW Release:	
Manufacturer:	Polycom Spectralink
Description:	The 8440 VoWiFi handset functions as a SIP device registered on the Hipath 4000.
Documentation:	
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 Spectralink** phones listed below with the **Siemens Hipath 400 V6** 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 codec's and DTMF
- Restart test
- Basic WLAN test were performed but are not part of this certification

Other scenarios, including data security/encryption and mass provisioning (via provisioning server) where not part of the certification.

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: August-October 2011

Test Duration:
August 25rd – September 2th, 2011
October 10th, 2011
November 17th, 2011

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

Test Personnel:
Siemens Hipath 4000:
Eddy Sterckx
E-Mail: eddy.sterckx@siemens-entreprice.com
Phone: +32.2.406 7197

Daniel Van Riet
E-Mail: daniel.vanriet@siemens-entreprice.com
Phone: +32 2 406 7336

Michel Lambrecht
E-Mail: michel.lambrecht@siemens-enterprise.com
phone: + 32 2 406 7325

Polycom:
Oliver Wick
Email: oliver.wick@polycom.com
Phone

Coordination:
De Braekeleer Eddy
Email: eddy.debraekeleer@siemens-entreprise.com
+32 2 406 7316

1.4 Test Results Summary

No major issues.
For details please have a look at the test results.

1.4.1 Problems

1. When the Polycom receives an SDP With attribute “inactive” then there is no Music on Hold heard. In the case the Polycom receives an “inactive” attribute, the Music on hold will not come from the PBX, but must be played back locally on the Polycom phone. Otherwise the end user will only hear silence during this hold scenario. Playing silence can confuse the end user, which might think the call is e.g. disconnected. So in this scenario, Polycom needs to play music on hold locally itself to the phone.

2. In some scenario’s Polycom puts the remote party on hold by sending the SDP attribute “SendOnly”. The PBX then answers with the attribute “RecvOnly”. This is fully RFC conform, and one of the most used scenario’s to signal a HOLD scenario. However in the case the Polycom sends the “SendOnly” attribute, and the PBX answers with the attribute “RecvOnly” in this case, the Polycom needs to stream a Music On hold inband. The Polycom does however sends no streaming at all in this case, and the remote user will hear silence while he is put on hold. Playing silence can confuse the end user, which might think the call is e.g. disconnected. Either the Polycom starts in this case with inband MOH streaming, or another possibility is that the Polycom signals a hold by sending the attribute “inactive” since in this case, it is the responsibility of the PBX to play music on hold inband to the remote party.

1.4.2 Restrictions

1. Not possible to configures “line” settings via WEB interface, data is not saved

2. U-APSD and WMM are required for the Spectralink 8440 phone and must be activated on the WLAN controller

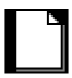
3. No display update on A-party in case of call transfer, call forwarding, call deflection. H4K Problem. Works as designed (Call transfer, Call forwarding or Call Deflect will not work and a HiPath 4000 CR (Change Request) would be needed. Statement confirmed by Mr Robert Stampfl , CP Development)

1.4.3 Remarks

1. In order to make the display update possible, the following has to be done. Change PRODE so that in PD07 (SBDSS1) the element Connected Number becomes mandatory for Outgoing Setup. CHANGE-PRODE:KIND=PD,PDNAME=PD07,SEC=WELMAND,SETNO=0,B22=10; CHANGE-PRODE:PD,PD07,ORG,,Y;

2. Device based forwarding is overruled if H4K based forwarding is configured.

3. The text in the GUI is not optimal for the HOLD mode. First of all the “new SDP type” will not ring any bell at all for any technician. “new SDP type” is a way too general description, and it also does not indicate that this setting handles about HOLD. Better would be to use a text like e.g. “Use media direction attributes for hold scenario’s (RFC3264)” or something similar.



D:\clearing HiPath 4000\Polycom HOLD p

4. The help text regarding the HOLD mode in the Web is not consistent. In the help text is written that RFC2543 is obsolete, which is true, but at the end the help text indicates that the default value is “disabled” which would mean that the obsolete RFC2543 is the default setting. This makes no sense as default value, since this is an obsolete RFC, and hold via RFC3264 is currently the best way to signal hold scenarios. So the remark about the fact that the obsolete RFC is the default value would

best be corrected. Note that hold via RFC2543 has the disadvantage that signalling an IP address 0.0.0.0 also makes that the RTCP cannot be signalled anymore, and also sometimes SBCs or SIP aware firewalls can have sometimes problems with the 0.0.0.0 IP address as hold indication.
see also http://www.ietf.org/rfc/rfc3264.txt
In this RFC of the year 2002 it was already stated that this method (at that time even) is not recommended anymore.
See paragraph 8.4 Putting a Unicast Media Stream on Hold
on page 17 :
RFC 2543 [10] specified that placing a user on hold was accomplished
by setting the connection address to 0.0.0.0. Its usage for putting
a call on hold is no longer recommended, since it doesn't allow for
RTCP to be used with held streams, doesn't work with IPv6, and breaks
with connection oriented media.

5. Polycom sends media attributes on SDP session level, and also on each stream level. Although this is most likely theoretical allowed, it can confuse technicians, since not everybody knows if the session level or the media level attribute has the highest priority. Normally, I would expect that the media attributes are either sent once on SDP session level, or sent on each media level. SDP example from Polycom :

```
v=0
o=- 1319026437 1319026438 IN IP4 10.10.102.159
s=Polycom IP Phone
c=IN IP4 10.10.102.159
t=0 0
a=sendonly
m=audio 2226 RTP/SAVP 8 18 127
a=sendonly
a=crypto:70 AES_CM_128_HMAC_SHA1_80 inline:oF5M94BGNgvZCbOlY5XiMRnENKpauI32Cl32uhcE|70:4
a=rtpmap:8 PCMA/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:127 telephone-event/8000
m=audio 2226 RTP/AVP 8 127
a=sendonly
a=rtpmap:8 PCMA/8000
a=rtpmap:127 telephone-event/8000
```

This issue can be seen in several Polycom LAN traces, here just an example :

D:\clearing HiPath 4000\Polycom HOLD p

6. With the same hold settings on the Polycom, we see in one single call where multiple hold/retrieve actions are executed, that the Polycom sometimes signals a hold via the 0.0.0.0 IP address, and sometimes Polycom signals a hold via the SDP Attribute “SendOnly”. It would be better that the Polycom signals a hold always via the same way, as chosen in the Polycom settings (Hold via RFC2543 or via RFC3264)

7. When the Polycom is set on hold by the PBX via the SDP attribute “SendOnly” and the Polycom answers with the SDP attribute “RecvOnly” then the PBX starts streaming MOH to the Polycom phone. This is fully RFC conform. Problem is however that the Polycom in this case goes to a U-APSD state (to save the battery?). Due to this, the phone does only “sporadic” contact the base station, but problem is that the phone gets each 20 milliseconds an RTP packet from the remote side with Music on Hold. In normal conditions, we get a ping reply of the phone in about 10 milliseconds. When the phone was set on hold, the ping reply is only received after several hundreds of milliseconds, sometimes even after 800 milliseconds. Due to the fact that the phone decides to go to the U-APSD (sleep) state which the RTP packets with music on hold are sent to the Polycom each 20 milliseconds, results towards the end user to playback of the music on hold with a very, very bad speech quality. The U-APSD sleep state is in this case to my opinion not optimal, since it degrades the played back music on hold tone a lot, and also this hold state will not occur that often, so most likely the battery time savings will be most likely minimal. Adviced is that the U-APSD state is not used in this specific hold state, since it degrades the MOH playback quality a lot.

8. Minor remark: Sometimes the Polycom phone sends RTP packets for x seconds, and then e.g. due to a hold, the Polycom does not send any RTP packets, but after the retrieval, RTP packets are again sent by the Polycom. In this last case, when the Polycom again starts to send RTP packets to the remote side, it would be advised that Polycom sets the RTP MAKER bit in the first sent RPT packet header after the inactivity. This will trigger the remote jitter buffer etc. Does not directly seem to give any functional problem, but is best practice to do so.

Here some background info :
What's the marker bit good for?

For voice packets, the marker bits indicates the beginning of a talkspurt.
Beginning of talkspurts are good opportunities to adjust the playout delay at the receiver to compensate for differences between the sender and receiver clock rates as well as changes in the network delay jitter. Packets during a talkspurt need to played out continuously, while listeners generally are not sensitive to slight variations in the durations of a pause.
The marker bit is a hint; the beginning of a talkspurt can also be computed by comparing the difference in timestamps and sequence numbers between two packets, assuming the timestamp clock rate is known.
Packets may arrive out of order, so that the packet with the marker bit is received after the second packet in the talkspurt. As long as the playout delay is longer than this reordering, the receiver can still perform delay adaptation.
If not, it simply has to wait for the next talkspurt.

9. When the Polycom receives an SDP attribute “inactive” then Polycom is responsible to play back MOH on the phone. In this case, there is however silence hear, which can confuse the end user which might think the call was disconnected.

10. The HOLD mode (HOLD via RFC2543 or via RFC3264) is not uniform described between the Web GUI and the phone settings. In the GUI is written “when enabled use SDP media direction attributes per RFC3264”. The same parameter is in the phone itself described as “RFC2543 Hold: yes/no” So answering with YES in the GUI enabled RFC3264, answering YES on the phone would according to the text enable RFC2543. Since the parameter is the same in both cases, one of the two descriptions is wrong.

11. When changing the Hold type in the GUI or in the phone, seems not to influence the way the Polycom signals his hold. LAN trace looks similar independent of the chosen hold type.

2 Configuration

2.1 Polycom Spectralink devices

- Handset 8440: V 4.0.0.20561

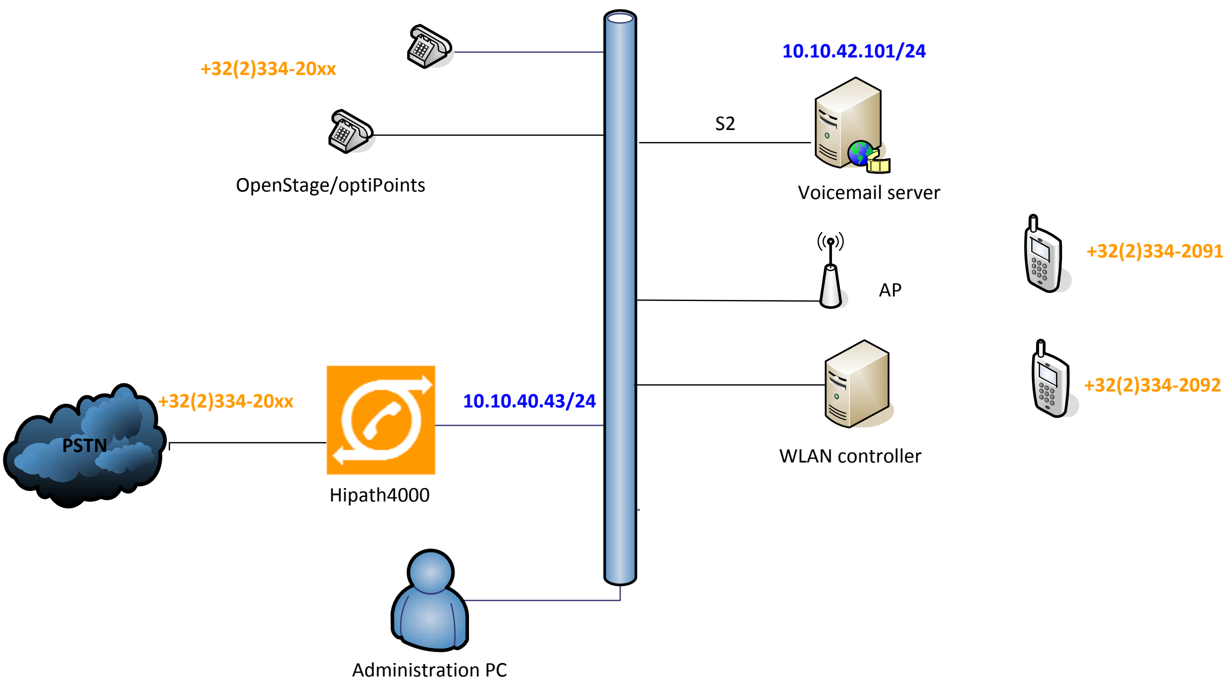
2.2 Hipath 4000

- RMX V6 R0.25.5
- Assistant V6 R1.10.0
- IP phones
    - OptiPoint 420
    - OpenStage 20/60
- STMI HG3500 pzksti40 38.001-007

2.3 Hipath Wireless Controller

- HW Version : C2400
- SW Version: 07.41.02.0009

2.4 Configuration Block Diagram



3 Test results in detail

The syntax of the abbreviations used in the test cases:

DUT = +3223342091 = DUT (device under test)
P1 = +3223342092 = **P**olycom Spectralink 8440 handsets
O1 = +3223342093 **O2** = +3223342094 = **O**penStage/optiPoint IP HFA phones
OS1 = +3223349440 = **O**penStage/optiPoint IP **S**IP phone
E1 = **E**xternal PSTN phone u

3.1 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 HiPath 4000 **registrar** information and verify that the phone registers	NOK	No effect when changing the line config parameters.
4	Change the HiPath 4000 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 via web GUI 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 (10.10.85.254).	OK	
7	The first node of the H4K is put out of service, which means that on the second node the backup registrar IP address is coming up.	NA	
8	The first node of the H4K is put in service again, which means that on the first node the registrar IP address is coming up.	NA	
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 hard phone and a soft client).	NA	No forking on H4K

3.2 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 P1. Verify that P1 is ringing (DUT receives ring back) and that the displays on the DUT and P1 show the correct called/calling number/name information. Subscriber P1 has special characters in the user name (éü…)	OK	
11	From the previous test case answer the call at P1 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 P1 answers. Verify that the DUT returns to idle state.	OK	
14	Initiate a call from P1 to the DUT. Verify that the DUT is ringing (P1 receives ringback) and that the displays on the DUT and P1 show the correct called/calling number/name information.	OK	
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	
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	

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	
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	

3.3 Telephony features

Test Case	Test Description	Result	Comment
28	Initiate a call from the DUT to internal subscriber P1. Answer the call at P1. Put the DUT **on hold** and verify that it receives Music-on-hold.	OK	No MOH
29	From the previous test case retrieve the DUT from hold and verify speech path between the DUT and P1.	OK	
30	Initiate a call from the DUT to internal subscriber P1. Answer the call at P1. Put the P1 **on hold** and verify that it receives Music-on-hold.	OK	Idem test 28
31	From the previous test case retrieve the P1 from hold and verify speech path between the DUT and P1.	OK	
32	Initiate a call from internal subscriber P1 to the DUT. Answer the call at the DUT. Put the DUT **on hold** and verify that it receives Music-on-hold.	OK	Idem test 28
33	From the previous test case retrieve the DUT from hold and verify speech path between the DUT and P1.	OK	
34	Initiate a call from internal subscriber P1 to the DUT. Answer the call at the DUT. . Put the P1 **on hold** and verify that it receives Music-on-hold.	OK	Idem test 28
35	From the previous test case retrieve P1 from hold and verify speech path between the DUT and P1.	OK	
36	Initiate a call from the DUT to internal subscriber O1. Answer the call at O1. Put the DUT **on consultation hold** and verify that it receives Music-on-hold.	OK	Interrupted MOH
37	From the previous test case retrieve the DUT from hold and verify speech path between the DUT and O1.	OK	
38	Initiate a call from the DUT to internal subscriber O1. Answer the call at O1. Put the O1 **on consultation hold** and verify that it receives Music-on-hold.	OK	Idem test 28
39	From the previous test case retrieve the O1 from hold and verify speech path between the DUT and O1.	OK	
40	Initiate a call from internal subscriber O1 to the DUT. Answer the call at the DUT. Put the DUT **on consultation hold** and verify that it receives Music-on-hold.	OK	Interrupted MOH

41	From the previous test case retrieve the DUT from hold and verify speech path between the DUT and O1.	OK	
42	Initiate a call from internal subscriber O1 to the DUT. Answer the call at the DUT. Put the O1 **on consultation hold** and verify that it receives Music-on-hold.	OK	Idem test 28
43	From the previous test case retrieve O1 from hold and verify speech path between the DUT and 01.	OK	
44	Initiate a call from the DUT to external subscriber E1. Answer the call at E1. Put the DUT **on consultation hold** and verify that it receives Music-on-hold.	OK	Interrupted MOH
45	From the previous test case return from hold and verify speech path between the DUT and E1.	OK	
46	Initiate a call from external subscriber E1 to the DUT . Answer the call at DUT. Put the E1 **on hold** and verify that it receives Music-on-hold.	OK	Idem test 28
47	From the previous test case return from hold and verify speech path between the DUT and E1.	OK	Idem test 28
48	Initiate a call from internal subscriber P1 to the DUT. Answer the call and initiate **consultation** at the DUT. Verify that P1 receives Music-on-hold while the DUT receives dial tone. Dial O1 at the DUT. Answer the call at O1. Verify that the DUT can **toggle** between P1 and O1.	OK	Idem test 28
49	Initiate a call from internal subscriber O1 to the DUT. Answer the call and initiate **consultation** at the DUT. Verify that O1 receives Music-on-hold while the DUT receives dial tone. Dial O2 at the DUT. Answer the call at O2. Verify that the DUT can **toggle** between O1 and O2.	OK	Idem test 28
50	Initiate a call from internal subscriber P1 to the DUT. Answer the call and make a **supervised transfer** at the DUT to O1. Verify that P1 receives Music-on-hold while the DUT receives dial tone.Verify that P1 and O1 have speech path, the displays are correct, and that the DUT returns to idle state.	OK	Idem test 28
51	Initiate a call from the DUT to internal subscriber O1. Answer the call and make a **supervised transfer** at the O1 to O2 so that the DUT and O2 are connected. Verify that the DUT and O2 have speech path, the displays are correct, and that the O1 returns to idle state.	OK	Idem test 28
52	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	Idem test 28

53	Initiate a call from the DUT to internal subscriber P1. Answer the call and initiate consultation at the DUT. Dial A2 and perform a **blind transfer** from A2 to P1. Answer P1 and verify that A2 and P1 have speech path, the displays are correct, and that the DUT returns to idle state.	NA	Blind transfer not supported on H4K
54	Initiate a call from the O1 to the DUT. Answer the call on the DUT. Perform a **ringing 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	
55	Initiate a call from the DUT to the O2. Answer the call on O2. Perform a **ringing 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	
56	Initiate a call from the internal subscriber O1 to the O2. Answer the call on O2. Perform a **ringing 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	
57	From the previous test case invoke the **last number redial** function on the DUT and verify that it calls O2.	OK	Last number redial via off hook key
58	Initiate a call to the DUT from an **external** subscriber E1. Answer the call, then disconnect. Verify that the external number can be called from the call history list.	OK	Call is performed but no number in call list only "RICHT" name. Performing a "edit number " then the number OK
59	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 P1. Verify that all parties have speech path and that the displays on the phones indicate the conference.	OK	Conference OK. But no display indication of conference on member site.
60	From the previous test case **release the conference master** (= DUT). Verify that the O1 and P1 are in two-party talk and the displays are updated accordingly.	**OK**	
61	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 P1. Verify that all parties have speech path and that the displays on the phones indicate the conference.	OK	Conference OK. But no display indication of conference on DUT and P1, on O1 OK Idem test 58
62	From the previous test case **release the conference master** (= O1). Verify that the DUT and P1 are in two-party talk and the displays are updated accordingly.	OK	No display update on DUT
63	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 display indication

64	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	DND activated via “feature setting” on the 8440. 486 Busy
65	Activate **call forwarding (CFU)** on the H4K to P1. Call the DUT from O1 and verify that the call is forwarded to P1 (H4K based forwarding).	OK	
66	Activate **call forwarding (CFNR)** on the H4K to P1. Call the DUT from O1 and verify that the call is forwarded to P1 (H4K based forwarding).	OK	
67	Activate **call forwarding (CFB)** on the H4K to P1. Call the DUT from O1 and verify that the call is forwarded to P1 (H4K based forwarding).	OK	Number off calls line1 reduced to 1.
68	Activate **call forwarding (CFU)** on the DUT to P1. Call the DUT from O1 and verify that the call is forwarded to P1 (device based forwarding).	OK	No display indication of CF on P1
69	From the previous test case invoke the **call forwarding (CFU)** function on the DUT to a external subscriber E1.	OK	
70	From the previous test case (68). Call the DUT from E1 and verify that the call is forwarded to P1	OK	No display indication of CF on P1
71	Activate **call forwarding (CFNR)** on the DUT to P1. Call the DUT from O1 and verify that the call is forwarded to P1 (device based forwarding).	OK	No display indication of CF on P1
72	From the previous test case invoke the **call forwarding (CNR)** function on the DUT to a external subscriber E1.	OK	
73	From the previous test case (XX). Call the DUT from E1 and verify that the call is forwarded to P1	OK	No display indication of CF on P1
74	Activate **call forwarding (CFB)** on the DUT to P1. Call the DUT from O1 and verify that the call is forwarded to P1 (device based forwarding).	OK	No display indication of CF on P1
75	From the previous test case invoke the **call forwarding (CFB)** function on the DUT to a external subscriber E1.	OK	
76	From the previous test case (XX). Call the DUT from E1 and verify that the call is forwarded to P1	OK	No display indication of CF on P1
77	Put the DUT and O1 in the same pickup group. Call O1 from P1. While O1 is ringing, dial the **Group Pick-up** code (*22) from the DUT and verify that speech path to P1 is established and the display shows correct caller information.	NA	Call pickup not supported for SIP on H4K
78	Call the DUT from O1. While connected, call the DUT from P1 and verify that a **call waiting** indication is presented on the DUT that shows the calling party information.	OK	Number off calls line1 reduced to 24

79	From the previous test case accept the waiting call and verify that speech path is established between the DUT and P1. Verify that O1 is put on hold.	OK	
80	O1 has **call waiting** disabled. O1 is on the call with O2 and the DUT tries to call O1.	OK	Number off calls line1 reduced to 1
81	Call OS1 from the DUT and **reject** the call at OS1. Verify that the DUT indicates the call rejection. OS1 SIP phone	OK	
82	Call OS1 from the DUT and **deflect** the call to O1. Verify that the DUT indicates the call deflection.	OK	No indication of deflection, in connect state display info OS1 on DUT!
83	Make the DUT busy and then call it from P1. 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	
84	From the previous test case retrieve the voicemail message and verify that the MWI is turned off.	OK	
85	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	
86	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	
87	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	
88	The DUT is put in an **HG** (hunt group) together with O1, and P1.	OK	
89	Large conference call between O1, E1, DUT, P1, A2 and OS1 (more than three parties involved). The conference initiator is O1.	OK	Large conference not supported for SIP on H4K Master conference HFA user
90	Call DUT from O1 and perform a **callback free** on O1. Check if after a call from DUT the callback is performed.	OK	
91	Make the DUT busy and then call it from O1 and perform a **callback Busy** on O1. Check if after DUT becomes free the callback is performed.	OK	In case of O1 OK In case of OS1 Not supported on H4K

3.4 Audio features

Test Case	Test Description	Result	Comment
92	Configure A3 to use the G.729A codec only. Call the DUT from A3 and verify that the connection is established with G.729A (use Wireshark).	OK	
93	Configure A3 to use the G.723 codec preferably. Call the DUT from A3 and verify that the connection is established with the first matching codec supported by the DUT or rejected if no match is found.	OK	Remark: the G.723 codec was tested with an optiPoint 420.
94	Configure the DUT for DTMF transmission via RFC 2833. Verify that from and to the DUT DMTF “telephony events” are sent.	OK	Traces taken by mirroring the STMI port
95	Configure the DUT for DTMF transmission via RFC 2833. Verify that the Xpression voicemail system can be accessed via DTMF.	OK	Traces taken by mirroring the STMI port
96	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

3.5 Restart test

Test Case	Test Description	Result	Comment
97	Unplug the STMI board and check if the phone indicates that it is out of service.	OK	After a few minutes “line unregistered”
98	Replug the STMI board and check if the phones register automaticaly	OK	
99	Reload of H4K and check if the phones register automatically	OK	

3.6 Interconnection with DAKS

Test Case	Test Description	Result	Comment
100		OK	

101		OK	

3.7 WLAN tests

In order to increase the compatibility between the Polycom Spectralink devices and the Siemens wireless network equipment (wireless controller and access points) some basic wireless tests were performed. Pure WLAN tests are not part of this certification.

4 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

5 Configuration Data

5.1 Hipath 4000

5.1.1 System Basics

```
    ADD-SBCSU:2091,FPP,SIP,1-1-8-
28,S0PP,75,75,6,6,6,6,0,0,N,0,0,,,"SBDSS1",Y,Y,0,10
    ,N,Y,,,5,0,1,,,"2091","2091",,,,,;

    ADD-SBCSU:2092,FPP,SIP,1-1-8-
29,S0PP,75,75,6,6,6,6,0,0,N,0,0,,,"SBDSS1",Y,Y,0,10
    ,N,N,,,5,0,,,,"2092","2092",,,,,;
```

5.2 Polycom

5.2.1 Documentation

http://support.polycom.com/PolycomService/support/us/support/voice/wi-fi/spectralink_8400_wireless.html

5.2.2 Basic Configuration

D:\Cert\
PolycomWLAN\8440.z

5.3 Wireless network

Wireless network settings (Siemens):

D:\Cert\
PolycomWLAN\C2400

5.4 H4K regen

D:\Cert\
PolycomWLAN\regen.

6 Confirmation

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

Oliver Wick	Eddy Sterckx,
POLYCOM	SEN Belgium