Certificate

OpenScape Ready

The connectivity of

Ascom i62

made by the company

Ascom Wireless Solutions

at the open interface SIP of

OpenScape Voice V5

has been certified in accordance with the test report dated 2011-11-22. The tests have been conducted conforming to DIN EN ISO 9001.

This certificate is only valid in conjunction with the full test report and the notes contained therein.

Siemens Enterprise Communications GmbH & Co. KG
Munich, 2011-12-12

Eddy De Braekeleer	Karl Donath
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OpenScape Ready

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Siemens Enterprise Communications

Test Report of Certification

ascom

Ascom i62 VoWiFi handset

with

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

Test Status: released
Release Date: November 22, 2011

Issued by: Siemens Enterprise Communications
D-81359 Munich

Owner: Eddy De Braekeleer
Department: SEN Service
Date: November, 2011
Document: Certification_ascom_H8KV5.doc

Contents:

History of Change

Date	***Description***	***Name***
April 2011	Initial Version	Eddy De Braekeleer SEN Service Customer Solution Lab Brussels E-Mail: eddy.debraekeleer@siemens-enterprise.com Phone: +32.2.406.7316
April 2011	Review	Karel Eeckelaert E-Mail: karel.eeckelaert@siemens-enterprise.com Phone: +32.2.406.7119
April 20 – 25 2011	Certification tests	Karel Eeckelaert E-Mail: karel.eeckelaert@siemens-enterprise.com Phone: +32.2.406.7119
May – November 2011	Additional testing & tracing for Ascom	Karel Eeckelaert E-Mail: karel.eeckelaert@siemens-enterprise.com Phone: +32.2.406.7119
November 2, 2011	Review test documents by ASCOM	Ascom Wireless Solutions: Matthew Williams Email: matthew.williams@ascom.se Phone +46 31 55 93 58
November 16, 2011	Release	Eddy De Braekeleer SEN Service Customer Solution Lab Brussels 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:	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: V7 41.02.0009

1.1.2 ASCOM VoWiFi

Certification:	Test of interface functionality between the OpenScape Voice and the Ascom i62 VoWiFi handset
Test Equipment:	OpenScape Voice in combination with an RG8702 (PRI) gateway and the Siemens HiPath Wireless C2400 Controller and Access Points (AP) 36XX
Initial software version i62	2.2.17
Recommended version i62	**2.3.16**
HW / FW Release:	902202[F2]
Manufacturer:	Ascom Wireless Solutions
Description:	The Ascom i62 VoWiFi handset functions as a SIP device registered on the OpenScape Voice server.
Documentation:	- Configuration Manual: Ascom i62 VoWiFi Handset
Test Network:	Test network of HiPath Ready Lab Brussels
Test Configuration:	See section 2.3

1.2 Test Strategy

This certification test for the **Ascom** 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

REMARKS:

Only basic WLAN test were performed – WLAN was not part of this certification. This part has been done by Enterasys & Ascom.
For succesful implementation it is necessary to even consider the WLAN test results and hints from the respective Ascom-WLAN Interoperability Tests and Certifications. They are available at Ascom for the Ascom i62 VoWiFi Handset with a number of different WLAN Infrastructures from several Vendors

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).

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.

1.2.1 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: April 2011

Test Duration: April 20th – 25th, 2011
Additional testing & tracing for Ascom between May and November 2011

Test Location: Siemens Enterprise Communications
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1.4 Test Results Summary

For details please have a look at the test results.

1.4.1 Problems

1.4.2 Restrictions

1) `The Ascom handsets ignore the Registration-expiry-timer sent in the 200-OK Message by the OpenScape Voice. Issue Resolved in Ascom Software 2.3.16`
2) The Ascom handsets do not send keep-alives, which results in the handset being disassociated from the Controller. This was resolved with controller version V7 31.xx. The controller now sends keep-alives to the Ascoms.
   The handsets are released with Controller V7 41.01 and up with Base-stations AP36XX.
3) Call Parking is currently not possible with the ASCOM devices.

1.4.3 Remarks

1) The Ascom i62 VoWiFi handset does provide only limited DNS-SRV functionality (not looking at the return list in the DNS response).
   In order to support "survivability routing"' the alternate routing (alternate proxy) possibility of the handset was used. The handset provides a list of IP addresses for registrar/proxy. The failover time to the second node was seen to be about 120 seconds and the failback time to be about 90 seconds (both times can vary a bit).
2) In the contact header of the REGISTER message there is the "expires" parameter. When the OpenScape Voice server acknowledges with a "200 OK" then the value of the "expires" parameter is one less.
   This does NOT cause the Ascom to break but a remark is made here on the compatibility of this with the RFC.

   `Ticket for OpenScape Voice will be written`
3) When using a service code on the Ascom handset for activating system forwarding on the OpenScape Voice server it does reply with "487 Request Terminated" which shows "Call Failed" on the Ascom handset.
   The reply "487 Request Terminated" is considered to be wrong as it is sent always and the functionality is not broken (why not send for example a CANCEL

?).
This is ONLY a display issue as the enabling of the forwarding via the service code does work for the Ascom handset.

`Ticket for OpenScape Voice will be written`

4) Compatibility of OpenScape Voice with RFC 3891 (replaces header)

`Question for OpenScape Voice`

5) The Ascom handsets do support server based forwarding (OpenScape Voice determines the forwarding via a service code) and phone based forwarding (the handset does invoke the forwarding, e.g. “302 Temporarily moved” message)

6) The OpenScape Voice server does not support overlap/post dialling by nature of the SIP protocol (block dialling).

7) When DTMF tones via RFC 2833 (telephony events) are sent towards the OpenStage phones, clicks are heard instead of tones. On the Ascom devices this does work correctly.

8) The local music on hold delivered by the Ascom handset is a tone (device based moH) but also the media server based moH is supported. The media server based moH (central) is the preferred one.

9) When the Ascom handset has setup a local three-party conference and leaves then the conference is ended. The joining of the other parties is not supported.

10) The display of the Ascom handset shows “no response” when receiving a “480 Temporarily not available” message. This is for example the case when the other party is refusing the call (Do not disturb).
This is ONLY a display issue as the functionality is working.

11) If an external call is transferred to Ascom through Attended Transfer, then the display of the Ascom does not update.

12) In previous Software versions, there was no possibility to disable call waiting. Since V2.3.14 this feature was added, and works as expected.

2 Configuration

2.1 Ascom devices

The Ascom devices have been configured by the Ascom personnel.

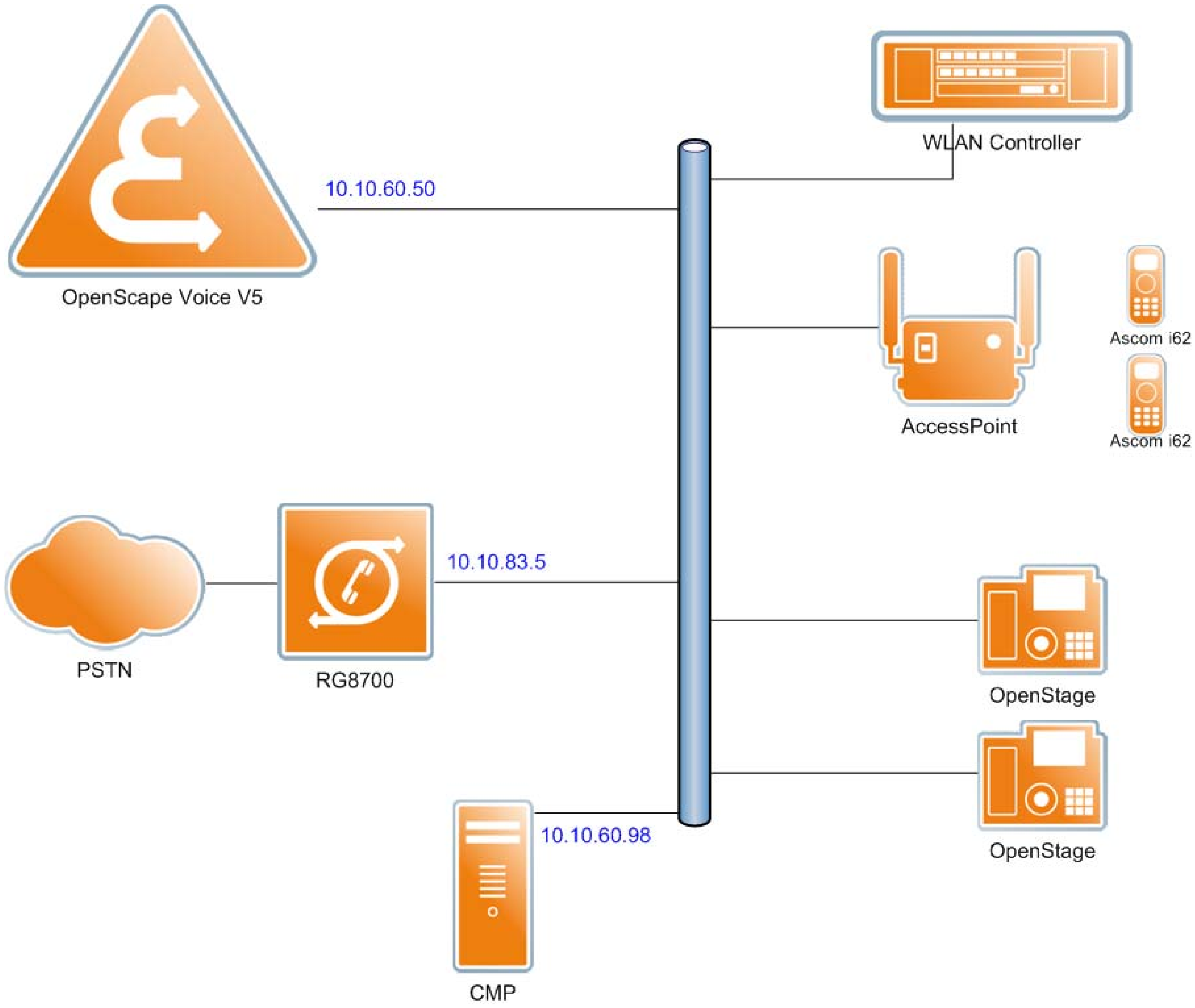
2.2 OpenScape Voice

- HW Version: Fujitsu Siemens RX330
- SW Version: V5.00.01.ALL.11_PS0017.E08
- IP phones
    - OptiPoint 420 V7 R2.2.0
    - 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: V7 41.02.0009
- Access Points: AP3610

2.4 Configuration Block Diagram



Ax = +3223342010/2016 = Ascom i62 handsets = DUT (device under test)
Oy = +3223342012/2013 = OpenStage/optiPoint 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 = +3223342010/2016 = Ascom i62 handsets = DUT (device under test)
Oy = +3223342012/2013 = OpenStage/optiPoint 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	For the GUI a proprietary admin interface is provided to be used via a password.
3	Program the phone via GUI with the OSV **registrar** information and verify that the phone registers	OK	The Ascom WinPDM program is used for that purpose. Remote management "over the air" provides a similar GUI.
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 via WinPDM 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 OpenScape Voice server is put out of service, which means that on the second node the backup registrar IP address is coming up.	OK	The Ascom handset does register on the second node and make calls via the second node. See remark 1 in section 1.4.3.
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	The Ascom handset does register on the first node and make calls via the first node. See remark 1 in section 1.4.3.
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 A3. Verify that A3 is ringing (DUT receives ring back) and that the displays on the DUT and A3 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 A3 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 A3 answers. Verify that the DUT returns to idle state.	OK	A3 shows missed call
14	Initiate a call from A3 to the DUT. Verify that the DUT is ringing (A3 receives ring back) and that the displays on the DUT and A3 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	The Ascom handset displays the indication that it is an external call (based on the configurable amount of digits)
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	The Ascom handset displays the indication that it is an external call (based on the configurable amount of digits)

3.1.3 Telephony features

Test Case	Test Description	Result	Comment
28	Initiate a call from the DUT to internal subscriber A3. Answer the call at A3. Put the DUT **on hold** and verify that it receives Music-on-hold.	OK	MOH delivered by MediaServer
29	From the previous test case retrieve the DUT from hold and verify speech path between the DUT and A3.	OK	MOH delivered by MediaServer
30	Initiate a call from internal subscriber A3 to the DUT. Answer the call at the DUT. From the DUT put A3 **on hold** and verify that it receives Music-on-hold.	OK	MOH delivered by MediaServer
31	Initiate a call from the DUT to internal subscriber A3. Answer the call at A3. Put the DUT **on hold** and verify that it receives Music-on-hold.	OK	MOH delivered by MediaServer

32	From the previous test case retrieve the DUT from hold and verify speech path between the DUT and A3.	OK	MOH delivered by MediaServer
33	Initiate a call from internal subscriber A3 to the DUT. Answer the call at the DUT. From the DUT put A3 **on hold** and verify that it receives Music-on-hold.	OK	MOH delivered by MediaServer
34	From the previous test case retrieve A3 from hold and verify speech path between the DUT and A3.	OK	MOH delivered by MediaServer
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	MOH delivered by MediaServer
36	From the previous test case retrieve the DUT from hold and verify speech path between the DUT and O1.	OK	MOH delivered by MediaServer
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	MOH delivered by MediaServer
38	From the previous test case retrieve the DUT from hold and verify speech path between the DUT and O1.	OK	MOH delivered by MediaServer
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	MOH delivered by MediaServer
40	From the previous test case retrieve A3 from hold and verify speech path between the DUT and A3.	OK	MOH delivered by MediaServer
41	Initiate a call from the DUT to internal subscriber A3. Answer the call and initiate **consultation** at A3. Verify that the DUT receives Music-on-hold.	OK	MOH delivered by MediaServer
42	From the previous test case return from consultation and verify speech path between the DUT and A3.	OK	
43	Initiate a call from internal subscriber A3 to the DUT. Answer the call and initiate **consultation** at the DUT. Verify that A3 receives Music-on-hold while the DUT receives dial tone.	OK	MOH delivered by MediaServer
44	From the previous test case dial O1 at the DUT. Answer the call at O1. Verify that the DUT can **toggle** between A3 and O1.	OK	
45	From the previous test case initiate a **supervised transfer** at the DUT so that A3 and O1 are connected. Verify that A3 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 O2. 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 A3. Answer the call and initiate consultation at the DUT. Dial A1 and perform a **blind transfer** from A3 to A1. Answer A1 and verify that A3 and A1 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	Only master shows conference, other parties show call to master
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.	* **X**	See remark 9.

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	Remark: the display of the DUT shows “no response” (translation of the “480 Temporarily not available”). See remark 10 in section 1.4.3
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	See remark 3
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	See remark 3
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	See remark 3
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	See remark 3
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 A3. While O1 is ringing, dial the **Group Pick-up** code (*7) from the DUT and verify that speech path to A3 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	On the DUT the message "user busy" is displayed.
71	Call O1 from the DUT and **reject** the call at O1. Verify that the DUT indicates the call rejection.	OK	On the DUT the message "user busy" is displayed.
72	Call O1 from the DUT and **deflect** the call to O2. Verify that the DUT indicates the call deflection.	OK	Display info un DUT gets update when O2 accepts the call.
73	Make the DUT busy and then call it from A3. 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	A recording sign is shown on the Ascom handset.
74	From the previous test case retrieve the voicemail message and verify that the MWI is turned off.	OK	The recording sign disappears from the Ascom handset.
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 A3 (more than three parties involved). The conference initiator is O1.	OK	
80	External party calls O1, O1 does blind transfer to DUT	OK	
81	External party calls O1, O1 does semi-attended transfer to DUT	OK	
82	External party calls O1, O1 does attended transfer to DUT	OK*	See remark 11
83	DUT has simultaneous ringing activated with O2. O1 calls DUT, both DUT and O2 ring.	OK	
84	O2 has simultaneous ringing activated with DUT. O1 calls O2, both DUT and O2 ring.	OK	

85	O1 is busy. DUT calls O1 and activates **CCBS**.	OK	
86	DUT calls O1. O1 does not respond. DUT activates **CCNR**.	OK	
87	DUT is busy. O1 calls DUT and activates **CCBS**.	OK	
88	O1 calls DUT. DUT does not respond. O1 activates **CCNR**.	OK	
89	DUT parks a call and retrieves a call.	NOK	See restriction 3.

3.1.4 Audio features

Test Case	Test Description	Result	Comment
90	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	
91	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. The G.723 codec is not supported in the i62 GUI but is in the supported codec list (SDP).
92	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 access point and wireless controller ports on the Ethernet switch.
93	Configure the DUT for DTMF transmission via RFC 2833. Verify that the Xpressions voicemail system can be accessed via DTMF.	OK	Traces taken by mirroring the access point and wireless controller ports on the Ethernet switch.
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

3.1.5 WLAN tests

In order to increase the compatibility between the Ascom devices and the Siemens wireless network equipment (wireless controller and access points) some basic wireless tests were performed. Please note that in order to use UAPSD, the Controller needs to be a least in V7.41.01.

3.1.6 Special Features

90	Firmware upgrade with WinPDM	OK	
91	Secure Voice (SRTP)	NA	
92	SIP over TCP (TSIP)	NA	
93	Secure Signalling (TLS)	NA	
94	Number of simultaneous calls can be configured from WinPDM.	OK	Configurable parameter: Device > Call > Busy on 1/Disable call waiting. Default value set to “No”. With this feature, a new incoming call can be rejected and busy indication sent back to the SIP proxy.
95			
96			

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
See attachment
"Ascom_i62_OSV_V5_Config_UserData_Export.zip"

4.2 Ascom VoWiFi

4.2.1 Documentation

User Manual Ascom i62 VoWiFi Handset (TD 92599GB)
Configuration Manual Ascom i62 VoWiFi Handset (TD 92675GB)
System Description Ascom VoWiFi System (TD 92313GB)
System Planning Ascom VoWiFi System (TD 92408GB)

Ascom's technical documentation is available through a local supplier.

4.2.2 Basic Configuration

Configuration of Ascom i62
See attached Ascom Number File
"Ascom.xcp"

Recommendations :

- **Ascom recommends that the parameter: Network > Network A > 802.11b/g/n channels should be set to "1,6,11" in conjunction with this WLAN infrastructure.**
- **The SIP Expiration value in the OSV should be set to 120 in stead of the standard value of 3600.**

5 Confirmation

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

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