Unify Ready

Technology connectivity certification

The connectivity of

Konftel 300 IP

developed by Konftel AB has been certified at the SIP Interface of OpenScape Voice V9 in accordance with the respective test report.

Konftel AB is now entitled to label the above mentioned product with the Unify Ready emblem.

The test was conducted conforming to DIN EN ISO 9001. This certificate is only valid in conjunction with the full test report and the notes contained therein. **Please consider that the test report only covers the functionality of the interface. The certificate and test report are not good for a statement of end-to-end functionality.**

Munich, November 24th 2016

Luzia Stephan

Director Technology Partner Program

UNIFY READY

UNIFY Harmonize your enterprise

Test Report of Certification

Konftel
300IP

with

OpenScape Voice Version 9

Test Status: Released
Release Date: 24/11/2016

Owner: Eddy Debraekeleer
Department: SER CM BELUX
Document: Certification_Konftel_300IP_OSV_V9_rev1.0

Contents:

History of Change **3**

1 Overview **4**

1.1 Test Object **4**

1.1.1 Basic Equipment 4
1.1.2 Konftel 4

1.2 Test Strategy **5**

1.2.1 Test Intensity 5
1.2.2 Measuring / Test Instruments 5

1.3 Realisation Data **5**

1.4 Test Results Summary **6**

1.4.1 Problems 6
1.4.2 Restrictions 6
1.4.3 Remarks 6

2 Configuration **7**

2.1 Konftel 300IP **7**

2.2 OpenScape Voice **7**

2.3 Configuration Block Diagram **7**

3 Test Results in Detail **8**

3.1 Connectivity and Basic Operation **8**

3.2 Basic call **9**

3.3 Telephony features **10**

3.4 Multi-account Support **12**

3.5 Audio features **13**

3.6 Remarks **13**

4 Configuration Data **14**

4.1 OpenScape Voice **14**

4.1.1 System Basics 14

4.2 Konftel **14**

4.2.1 Documentation 14
4.2.2 Basic Configuration 14

5 Confirmation **14**

History of Change

Date	***Description***	***Name***
June 2016	Initial Version	Eddy Sterckx Email: eddy.sterckx@unify.com Phone: +32 2 406 7179
9-10 June 2016	Certification tests	Michel Lambrecht Email: michel.lambrecht@unify.com Phone: +32 2 406 7325 Eddy Sterckx Email: eddy.sterckx@unify.com Phone: +32 2 406 7179
10 June 2016 14 June 2016	Review test document & update results Review test document	Eddy Sterckx Email: eddy.sterckx@unify.com Phone: +32 2 406 7179 Eddy Sterckx Email: eddy.sterckx@unify.com Phone: +32 2 406 7179
24 November 2016	Final review	Eddy De Braekeleer UNIFY Service PS E-Mail: eddy.debraekeleer@unify.com Phone: +32.2.406 7316
24 November 2016	Release	Eddy De Braekeleer UNIFY Service PS E-Mail: eddy.debraekeleer@unify.com Phone: +32.2.406 7316

1 Overview

1.1 Test Object

1.1.1 Basic Equipment

Test system: **OpenScape Voice**

Software Version:
- Switch Type: Simplex
- Software Version V9 R0.6.2

Gateways

1.1.2 Konftel

Certification: Test the functionality of the Konftel 300IP with the OpenScape Voice

Test Equipment: OpenScape Voice, Openstage SIP, Konftel 300IP

Software Release: 2.5.13

HW / FW Release: Konftel 300IP

Manufacturer: Konftel

Description: Konftel 300IP functions as a SIP device registered on the OSV.

Documentation: Installation Guide

Test Network: Test network of OpenScape Ready Lab Brussels

Test Configuration: See section 2.3

1.2 Test Strategy

This certification test for the **Konftel** 300IP with the **Unify OpenScape Voice** focused on the verification of the SIP interface in the following scenarios:

- Basic phone configuration and registration
- Basic calls
- Telephony feature verification
- Multi account support
- Audio features, including codec's and DTMF

Following topics were not part of the certification:

- security/encryption (802.1x)
- 802.1q (vlan tagging)
- Factory reset

1.2.1 Test Intensity

Scope of the tests is to execute and 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 a 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.
Unify Communications therefore assumes no responsibility for the compliance to these requirements.

1.2.2 Measuring / Test Instruments

No special hardware for tracing. Tracing was done on the switch equipment (mirroring ports, Wireshark) and on the OpenScape Voice server.

1.3 Realisation Data

Test Preparation: June 9th 2016

Test Duration: June 9-10th 2016

Test Location: Unify Communications
Demeurslaan 134
1654 Huizingen
International Solution Lab

Test Personnel: Unify:
Michel Lambrecht
Email: michel.lambrecht@unify.com

Eddy Sterckx
Email: eddy.sterckx@unify.com

Coordination: Eddy De Braekeleer
E-Mail: eddy.debraekeleer@unify.com

1.4 Test Results Summary

For the details please have a look at the test results.

1.4.1 Problems

1
2
3
4

1.4.2 Restrictions

1 Test case 3.2.1&2 Functionality is OK but only the “from” extension number and not the “from” name is shown in ringing and connect state.

2 Test case 3.2.5&6 Functionality is OK but in ringing state only the “from” extension name is shown and in connected state only the “from” extension number.

3 Test case 3.2.10 Functionality is OK but in ringing state only the “from” name is shown and in connected state only the “from” extension number.

4 Test case 3.3.1 & 3.3.5 In the SIP/SDP ACK from OSV to DUT a “Sendonly” is send, DUT replays with a 200 ok SIP/SDP “inactive” and not “reconly” this result in no MOH is played on DUT. DUT on the other hand indicates “HELD” on the display.

5 Test case 3.3.19 Functionality is OK but no display update on DUT

6 Test case 3.3.23 Functionality is OK but no display update in ringing nor connected state on DUT

1.4.3 Remarks

1 Test case 3.3.13&15 only on the Master conference device shows ‘Active Conference info’ on display. W.a.d. on OSV.

2
3
4

2 Configuration

2.1 Konftel 300IP

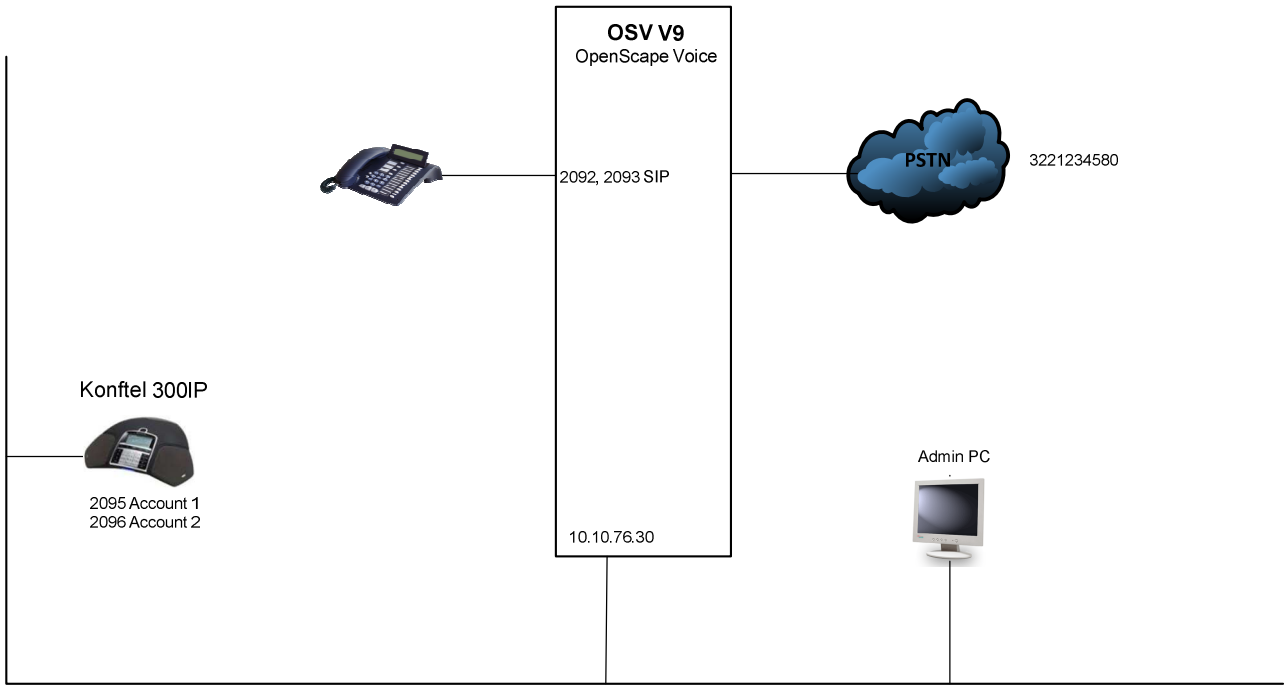
The Konftel 300IP device has been configured by Unify.

2.2 OpenScape Voice

- Switch Type: Simplex
- SW Version : V9 R0.6.2

  Openstage SIP V3 R4.8.0

2.3 Configuration Block Diagram



3 Test Results in Detail

3.1 Connectivity and Basic Operation

The following table shows the list of test cases and the associated results for the verification of basic phone connectivity via the SIP interface.

Test Case	Test Description	Result	Comment
		300IP	
3.1.1	Connect the test phone to its AC power supply and the LAN. Verify that the phone obtains a valid IP address from the **DHCP** server.	OK	
3.1.2	Connect the test phone to a **PoE** source and verify that it powers up correctly and obtains a valid IP address from the **DHCP** server.	OK	
3.1.3	Connect a PC to the lab LAN and verify that access to the web GUI of the test phone is possible. **User ID=** '**Admin'**, **password=** '**1234'**.	OK	Use http://
3.1.4	Program the phone via web GUI with the OpenScape Voice **registrar** information and verify that the phone registers.	OK	
3.1.5	Change the OpenScape Voice subscriber settings so that **Digest Authentication** is required for the registration. Verify that the phone does not register.	OK	
3.1.6	Add the information for **Digest Authentication** to the test phone settings via web GUI and verify that the phone registers.	OK	
3.1.7	Verify that the test phone displays the local date and time correctly that is provided by the SNTP server.	OK	

3.2 Basic call

The following table shows the list of test cases and the associated results for the verification of basic calls. All tests are executed with G.711A codec setting.

Test Case	Test Description	Result	Comment
		300IP	
3.2.1	Initiate a call from the DUT to **internal** subscriber 2092. Verify that 2092 is ringing (DUT receives ring back) and that the displays on the DUT and 2092 show the correct called/calling number/name information.	NOK	See 1 in section 1.4.2
3.2.2	From the previous test case answer the call at 2092 and verify speech path between both phones. Verify that the phone displays show the correct information after the call connected.	NOK	See 1 in section 1.4.2
3.2.3	From the previous test case **disconnect** the call at the DUT and verify that both phones return to idle state.	OK	
3.2.4	Repeat the previous call, but disconnect the DUT before 2092 answers. Verify that the DUT returns to idle state.	OK	
3.2.5	Initiate a call from 2092 to the DUT. Verify that the DUT is ringing (2092 receives ring back) and that the displays on the DUT and 2092 show the correct called/calling number/name information.	NOK	See 2 in section 1.4.2
3.2.6	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.	NOK	See 2 in section 1.4.2
3.2.7	From the previous test case disconnect the call at the DUT and verify that both phones return to idle state.	OK	
3.2.8	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	
3.2.9	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	
3.2.10	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.	NOK	See 3 in section 1.4.2
3.2.11	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

The following table shows the list of test cases and the associated results for the verification of telephony features in various call situations. All tests are executed with G.711A codec.

Test Case	Test Description	Result	Comment
		300IP	
3.3.1	Initiate a call from the DUT to internal subscriber 2092. Answer the call at 2092. Put the DUT **on hold** and verify that it receives Music-on-hold.	NOK	DUT indicates 'HELD" on display See 4 section 1.4.2
3.3.2	From the previous test case retrieve the DUT from hold and verify speech path between the DUT and 2092.	OK	
3.3.3	Initiate a call from internal subscriber 2092 to the DUT. Answer the call at the DUT. From the DUT put 2092 **on hold** and verify that it receives Music-on-hold.	OK	DUT indicates 'HOLD" on display
3.3.4	From the previous test case retrieve 2092 from hold and verify speech path between the DUT and 2092.	OK	
3.3.5	Initiate a call from the DUT to internal subscriber 2092. Answer the call and initiate **consultation** at 2092. Verify that the DUT receives Music-on-hold.	NOK	DUT indicates 'HELD" on display See 4 section 1.4.2
3.3.6	From the previous test case return from consultation and verify speech path between the DUT and 2092.	OK	
3.3.7	Initiate a call from internal subscriber 2092 to the DUT. Answer the call and initiate **consultation** at the DUT. Verify that 2092 receives Music-on-hold while the DUT receives dial tone.	OK	DUT indicates 'HOLD" on display
3.3.8	From the previous test case dial 2093 at the DUT. Answer the call at 2093. Verify that the DUT can **toggle** between 2092 and 2093.	NA	No toggle function on DUT
3.3.9	From the previous test case initiate a **supervised transfer** at the DUT so that 2092 and 2093 are connected. Verify that 2092 and 2093 have speech path, the displays are correct, and that the DUT returns to idle state.	NA	No supervised transfer function on DUT
3.3.10	Initiate a call from the DUT to internal subscriber 2092. Answer the call and initiate consultation at the DUT. Dial 2093 and perform a **blind transfer** from 2092 to 2093. Answer 2093 and verify that 2092 and 2093 have speech path, the displays are correct, and that the DUT returns to idle state.	OK	
3.3.11	From the previous test case (3.3.9) invoke the **last number redial** function on the DUT and verify that it calls 2093.	OK	
3.3.12	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	
3.3.13	Initiate a call from the DUT to internal subscriber 2092. Answer the call and initiate a **three-way conference** from the DUT with 2093. Verify that all parties have speech path and that the displays on the phones indicate the conference.	OK	Only the DUT display shows 'Active Conference info'. See 1 section 1.4.3

Test Case	Test Description	Result	Comment
		300IP	
3.3.14	From the previous test case go on-hook at 2092. Verify that the DUT and 2093 are in two-party talk and the displays are updated accordingly.	OK	
3.3.15	Call an external number from 2092. Once connected use 2092 to add the DUT to a **three-way conference**. Verify that all parties have speech path and that the displays on the phones indicate the conference.	OK	Only 2092 display shows 'Active Conference info'. See 1 section 1.4.3
3.3.16	From the previous test case **release the conference master** 2092. Verify that the DUT and the external phone disconnected.	OK	
3.3.17	Call the DUT from 2092 after the **Do-Not-Disturb** function was activated. Verify that the call is rejected.	NA	No DND function on DUT
3.3.18	Activate **call forwarding** on the DUT to 2093. Call the DUT from 2092 and verify that the call is forwarded to 2093.	NA	No call forwarding function on DUT
3.3.19	Put the DUT and 2092 in the same pickup group. Call 2092 from 2093. While 2092 is ringing, dial the **Pick-up** code (*7) from the DUT and verify that speech path to 2093 is established and the display shows correct caller information.	NOK	See 5 in section 1.4.2
3.3.20	Call the DUT from 2092. While connected, call the DUT from 2093 and verify that a **call waiting** indication is presented on the DUT that shows the calling party information.	OK	
3.3.21	From the previous test case accept the waiting call and verify that speech path is established between the DUT and 2093. Verify that 2092 is put on hold.	OK	
3.3.22	Call 2092 from the DUT and **reject** the call at 2092.	OK	
3.3.23	Call 2092 from the DUT and **deflect** the call to 2014. Verify that the DUT indicates the call deflection.	NOK	See 6 in section 1.4.2
3.3.24	Make the DUT busy and then call it from 2092. Verify that the call is forwarded to the voicemail system (Xpressions) and that the **message waiting indication** (**MWI**) on the DUT is turned on.	NA	No MWI/VM function on DUT
3.3.25	From the previous test case retrieve the voicemail message and verify that the MWI is turned off.	NA	
3.3.26	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.	NA	
3.3.27	While the MWI is lit on the DUT, reboot the Xpressions server. Verify that after the reboot is complete, the MWI is turned on.	NA	

3.4 Multi-account Support

The following table shows the list of test cases and the associated results for the verification of multi account support of the **Konftel** SIP phones.

Test Case	Test Description	Result	Comment
		300IP	
3.4.1	Program the second phone (account) via web GUI with the OpenScape Voice **registrar** information and verify that the phone registers.	OK	
3.4.2	Change the OpenScape Voice subscriber settings so that **Digest Authentication** is required for the registration. Verify that the phone does not register.	OK	
3.4.3	Add the information for **Digest Authentication** to the test phone settings via web GUI and verify that the phone registers.	OK	
3.4.4	Set the second account as “default account”, check if DUT indicates that this account is active.	OK	
3.4.5	Initiate a call from the DUT to **internal** subscriber 2092. Verify that 2092 is ringing (DUT receives ring back) and that the displays on 2092 show the correct called/calling number/name information.	OK	
3.4.6	From the previous test case answer the call at 2092 and verify speech path between both phones	OK	
3.4.7	Initiate a call from 2092 to the DUT account 1. Verify that the DUT is ringing (2092 receives ring back). Answer the call and verify speech path between both phones.	OK	
3.4.8	From DUT, setup a five-way conference, two outgoing call from account 2 and two incoming calls to account 1. Verify that the conference can be established	OK	

3.5 Audio features

The following table shows the list of test cases and the associated results for the verification of various audio features, including codec settings and DTMF transmission.

Test Case	Test Description	Result	Comment
		300IP	
3.5.1	Configure 2092 to use the G.729 codec only. Call the DUT and verify that the connection is established with G.729 (use Wireshark).	OK	
3.5.2	Configure 2092 to use the G.729 codec preferably. Call the DUT and verify that the connection is established with the first matching codec supported by the DUT or rejected if no match is found. DUT G711 only	OK	2092 G.729 only, call rejected 2092 G.729 and G.711, call G.711
3.5.3	Configure 2092 to use the G.722 codec preferably. Call the DUT and verify that the connection is established with G.722 (use Wireshark).	OK	
3.5.4	Configure the DUT for DTMF transmission via RFC 2833. Call the Auto Attendant (Conf bridge) from the DUT. Verify that the Auto Attendant responds to the phone's DTMF keys (also use Wireshark).	OK	
3.5.5	Configure the DUT for DTMF transmission via inband tones. Call the Auto Attendant (Conf bridge) from the DUT. Verify that the Auto Attendant responds to the phone's DTMF keys.	OK	

3.6 Remarks

Meanings of Abbreviations:

OK	Testcase successful
NOK	Testcase NOT successful
NA	Testcase not applicable
NP	Testcase 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

4 Configuration Data

4.1 OpenScape Voice

4.1.1 System Basics

The OpenScape Voice configuration (screenshots).

DUT, Subscriber 2095 account 1, with digest authentication.


2095_digest-auth.zip

DUT, Subscriber 2095 account 1, without digest authentication.


2095_no_digest_auth.zip

4.2 Konftel

4.2.1 Documentation

*www.**konftel**.com*

4.2.2 Basic Configuration

Konftel 300IP configuration (screenshots):


Config.zip

5 Confirmation

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

Konftel

Eddy Sterckx, Michel Lambrecht
Unify Belgium