Alcatel Lucent Application Partner Program
Inter-Working Report

Partner: Polycom
Application type: SIP Phone
Application name: SoundPoint IP 670
Alcatel-Lucent Platform: OmniPCX Office

Certification overview

Date of the certification	February 2011

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Alcatel-Lucent Communication Platform	OmniPCX Office
Alcatel-Lucent Communication Platform Release	R810 / 049.004
AAPP member application version	SoundPoint IP 670 Rev 3.3.1.0769
Application Category	Terminals

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Revision History

Edition 1: creation of the document – *February 2011*
Edition 2: extension to OXO R9.0 – *January 2013*

Test results

☐ Passed ☐ Refused ☐ Postponed

☑ Passed with restrictions

Refer to the section **Erreur ! Source du renvoi introuvable.** for a summary of the test results.

IWR validity extension

The validity of this IWR has been extended to the following software releases/products:

- OmniPCX Office Release 9.0 – *January 2013*

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1 Introduction

This document is the result of the certification tests performed between the AAPP member's application and Alcatel-Lucent's platform.

It certifies proper inter-working with the AAPP member's application.

Information contained in this document is believed to be accurate and reliable at the time of printing. However, due to ongoing product improvements and revisions, Alcatel-Lucent cannot guarantee accuracy of printed material after the date of certification nor can it accept responsibility for errors or omissions. Updates to this document can be viewed by Business Partners on the Technical Support page of the Enterprise Business Portal (https://businessportal.alcatel-lucent.com) in the Application Partner Interworking Reports corner.

2 Validity of the Interworking Report

This Interworking report specifies the products and releases which have been certified.

This inter-working report is valid unless specified until the AAPP member issues a new major release of such product (incorporating new features or functionalities), or until Alcatel-Lucent issues a new major release of such Alcatel-Lucent product (incorporating new features or functionalities), whichever first occurs.

A new release is identified as following:

- a Major Release" is any x. enumerated release. Example Product 1.0 is a major product release.
- a "Minor Release" is any x.y enumerated release. Example Product 1.1 is a minor product release

The validity of the Interworking report can be extended to upper major releases, if for example the interface didn't evolve, or to other products of the same family range. Please refer to the "IWR validity extension" chapter at the beginning of the report.

? ***Note:*** *The Interworking report becomes automatically obsolete when the mentioned product releases are end of life.*

3 Limits of the Technical support

Technical support will be provided only in case of a valid Interworking Report (see chapter 2 “Validity of the Interworking Report) and in the scope of the features which have been certified. That scope is defined by the Interworking report via the tests cases which have been performed, the conditions and the perimeter of the testing as well as the observed limitations. All this being documented in the IWR. The certification does not verify the functional achievement of the AAPP member’s application as well as it does not cover load capacity checks, race conditions and generally speaking any real customer's site conditions.

Any possible issue will require first to be addressed and analyzed by the AAPP member before being escalated to Alcatel-Lucent.

For any request outside the scope of this IWR, Alcatel-Lucent offers the “On Demand Diagnostic” service where assistance will be provided against payment.

For more details, please refer to Appendix F “AAPP Escalation Process”.

3.1 Case of additional Third party applications

In case at a customer site an additional third party application NOT provided by Alcatel-Lucent is included in the solution between the certified Alcatel-Lucent and AAPP member products such as a Session Border Controller or a firewall for example, Alcatel-Lucent will consider that situation as to that where no IWR exists. Alcatel-Lucent will handle this situation accordingly (for more details, please refer to Appendix F “AAPP Escalation Process”).

4 Application information

Application type: VoIP SIP Phone

Application commercial name: Polycom SoundPoint IP 670

Application version: Rev 3.3.1.0769 / BootROM 4.3.0.0246

Interface type : SIP / IP / Ethernet

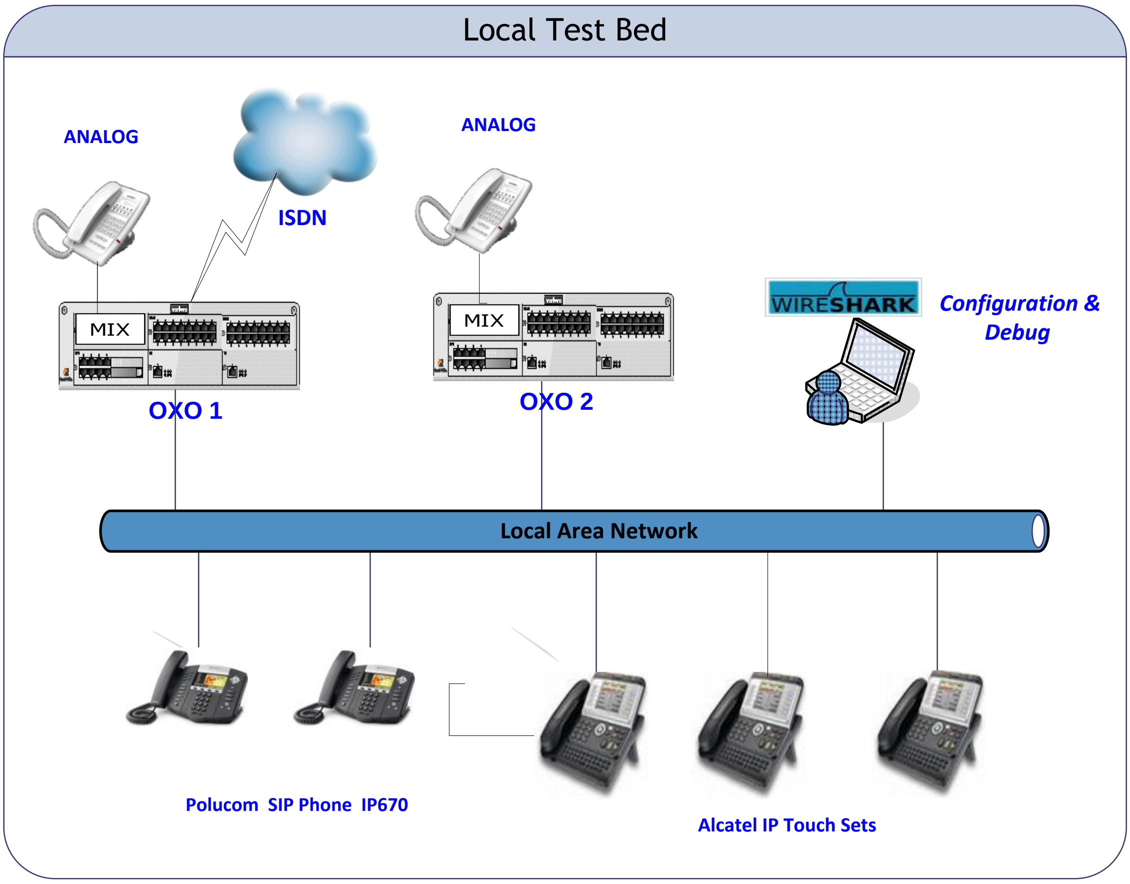
Interface version (if relevant):

Brief application description *(From Polycom SoundPoint IP 670 data sheet)* :

The SoundPoint IP 670 is a premium SIP desktop phone with color display that delivers a rich voice, visual, and applications experience.

- A large, vibrant color display for easier viewing and navigation
- Revolutionary voice quality with Polycom HD Voice™ technology
- Integrated Gigabit Ethernet (GigE) switch to enable bandwidth-intensive applications
- Six lines in stand-alone mode and 34 lines with three Polycom SoundPoint IP Color Expansion Modules
- A visually improved platform to enable a high level of integration with productivity-enhancing applications and business processes
- Built-in USB port to support applications, such as Local Call Recording from the Polycom Productivity Suite

5 Test Environment



5.1 Hardware configuration

Alcatel-Lucent Communication Platform:

- OmniPCX Office Rack
- PowerCPU

Setup Details:

Setup Information	
Module	**Details**
OXO 1 IP address	192.168.92.246 /24
Voicemail No	500
Attendant No	100
OXO Extension Details used for test	
IP Touch numbers	134, 135 & 136
SoundPoint IP 670	141 & 145
x9 series Set No	101, 102
OXO 2 IP address	192.168.91.246 /24
SoundPoint IP 670	241
IP Touch numbers	240 & 242
x9 series Set No	202

Note: the two OmniPCX Office systems are connected via private SIP trunking.

5.2 Software configuration

- **Alcatel-Lucent Communication Platform**: OmniPCX Office R810/049.004
- **Partner Application**: SoundPoint IP 670 Rev 3.3.1.0769 - BootROM 4.3.0.0246

Note: SoundPoint IP 670 is registered in the OmniPCX Office as “Open SIP phone”.

6 Summary of test results

6.1 Summary of main functions supported

Features	Status	Comments
Initialization including network configuration	OK	
SIP registration	OK	DHCP is restricted to ALU IP Phones
SIP authentication	OK	
Voice over IP and RTP codec support	OK	
Outgoing Call	OK	
Incoming Call	Ok_But	Feature is activated and working but wrong feedback is sent to the SIP user
Features During Conversation	OK	Only available from device (local)
Call Transfer	OK	Semi – Attended / Unattended (Blind)Transfer is not supported for SIP sets on OXO
Attendant	OK	
Voice mail interaction and indication	OK	

6.2 Summary of problems

- Call forward activated in call server via prefix is working but wrong acknowledgement message is send to SIP device - crms00351769.
- Call forward on busy is not working for SIP phones.
- Appointment configuration in call server is not working.

6.3 Summary of limitations

- DHCP mode is restricted to ALU IP Phones.
- There is no count of new voice mail messages on the display.
- In Conference we are unable to see the other user information other than the one who initiated the conference.
- G723 is not supported on SoundPoint IP 670.
- Call feature activation in the call server (eg CFU/CFB) is not displayed on the SIP device.
- Semi - Attended and blind Transfer is not supported in OXO

6.4 Notes, remarks

- SoundPoint IP 670 is registered in the OmniPCX Office as “Open SIP phone”.

7 Test Result Template

The results are presented as indicated in the example below:

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Test case 1** • Action • Expected result	☐	☒	☐	
2	**Test case 2** • Action • Expected result	☐	☒	☐	The application waits for PBX timer or phone set hangs up
3	**Test case 3** • Action • Expected result	☒	☐	☐	Relevant only if the CTI interface is a direct CSTA link
4	**Test case 4** • Action • Expected result	☐	☐	☒	No indication, no error message
…	…	☐	☐	☐	

Test Case Id: a feature testing may comprise multiple steps depending on its complexity. Each step has to be completed successfully in order to conform to the test.
Test Case: describes the test case with the detail of the main steps to be executed the and the expected result
N/A: when checked, means the test case is not applicable in the scope of the application
OK: when checked, means the test case performs as expected
NOK: when checked, means the test case has failed. In that case, describe in the field "Comment" the reason for the failure and the reference number of the issue either on Alcatel-Lucent side or on Application Partner side
Comment: to be filled in with any relevant comment. Mandatory in case a test has failed especially the reference number of the issue.

8 Test Results

8.1 Connectivity and Setup

8.1.1 Test Objectives

These tests shall verify that the different components are properly connected and can communicate together (the external application and the Alcatel-Lucent Communication Platform is connected and the interface link is operational).

8.1.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**SIP sets registration to OXO in static IP addressing** Create the IP users 141 & 145 on OXO with Open SIP phone profile. Configure the SIP phones 141 & 145 with OXO 1 CPU IP@ as SIP Register IP address. Check the phone registration and display. Note: SIP authentication is disabled for these users, the password doesn't matter.	☐	☒	☐	
2	**SIP sets registration to OXO in static IP addressing** For this test we will try to register the SIP phone with authentication enabled. Configure the SIP phones 141 & 145 with SIP Register IP address = OXO 1 CPU IP@ and SIP authentication password in the OXO. Check the phone registration and display. After make the same actions with a wrong password and check that the phone is rejected.	☐	☒	☐	
3	**DHCP registration with OXO internal DHCP server**	☒	☐	☐	Restricted to ALU IP extensions
4	**NTP registration** The SIP phone 141 & 145 are configured to retrieve the date and time from the OXO IP address. Check the phone retrieves the right date and time information and displays it.	☐	☒	☐	
5	**Support of "423 Interval Too Brief" (1)** The SIP phone 141 is configured with a value lower than 120 seconds. Check the phone registration and display	☐	☒	☐	
6	**Signaling TCP-UDP.** If applicable configure your SIP set to use the	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
	protocol SIP over UDP and TCP In the two cases, check the registration and basic calls.				

8.2 Audio codec negotiations/ VAD / Framing

8.2.1 Test Objectives

These tests check that the phones are using the configured audio parameters (codec, VAD, framing).

Phone configuration :
Configure TB 30 to use G.722, G.711 A-law, G.711 µ-law, G.729, G.723 in this order (unless otherwise stated).
Configure IP Touch with codec G.711 and to NOT use VAD (unless otherwise stated).

8.2.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Select G711 A-law as 1st codec in 141 Call from 141 **to** 134 Check that the call is established in G711 A-law. Check audio quality Call from 134 **to** 141 Check that the call is established in G711 A-law. Check audio quality	☐	☒	☐	
2	Select G729 as 1st codec in 141 Call from 141 **to** 134 Check that the call is established in G729 Check audio quality Call from 134 **to** 141 Check that the call is established in G729 Check audio quality	☐	☒	☐	
3	Select G723 as 1st codec in 141 Call from 141 **to** 134 (enable only G723 in 141) Check that the call is established in G729 Check audio quality Call from 134 **to** 141 Check that the call is established in G729 Check audio quality.	☒	☐	☐	G723 is not support on SoundPoint IP 670
4	**Configure 141 to use VAD** **Configure 134 NOT to use VAD** **Call from 141 to 134**	☐	☒	☐	VAD enabled on SoundPoint 670 : not tested

	Check that the call is established in G711 A-law. Check audio quality **Configure 141 to use VAD** **Configure 134 to use VAD** Redo the same tests **Configure 141 NOT to use VAD** **Configure 134 to use VAD** Redo the same tests				
5	**In OXO enable codec pass through for SIP phones** **Call from 141 to 145** Check that the call is established using G.722 Check audio quality	☐	☒	☐	
6	**In OXO 1 and OXO 2 enable codec pass through for SIP phone ; direct RTP and codec pass through for SIP trunk. G722 is preferred codec in SoundPoint IP 670.** **Call from 141 to 241** Check that the call is established using direct RTP in G722. Check audio quality	☐	☒	☐	
7	**In OXO 1 and OXO 2 disable direct RTP and codec pass through for SIP trunk ARS table is configured with "default" codec. G722 is preferred codec in SoundPoint IP 670.** **Call from 141 to 241** Check that the call is established in G729. Check audio quality.	☐	☒	☐	
8	**In OXO 1 and OXO 2 disable direct RTP and codec pass through for SIP trunk** **ARS table is configured with codec G711 - 30ms** **Call from 141 to 241** Check that the call is established in G711. Check audio quality	☐	☒	☐	

8.3 Outgoing Calls

8.3.1 Test Objectives

The calls are generated to several users belonging to the same network.
Called party can be in different states: free, busy, out of service, do not disturb, etc.
Calls to data devices are refused.
Points to be checked: tones, voice during the conversation, display (on caller and called party), hang-up phase.
OXO prefixes are mandatory for several tests of this section. For more information refer to the appendix E.

Note: dialing will be based on direct dialing number but also using programming numbers on the SIP phone (if available).

8.3.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Call to a local user** With SIP Phone 141 call the OXO IP Touch 134. Check that 134 is ringing. Take the call and check ring back tone audio and display.	☐	☒	☐	
2	**Call to local user with no answer** With SIP Phone 141 call the IP Touch 134. And never take the call. Check time out and display. Note that 134 don't have a Voice Mail	☐	☒	☐	Call is automatically cancelled by SoundPoint IP 670 after 1 mn.
3	**Call to another SIP set** With the SIP phone 141 call the other SIP Phone 145 Check the display and audio during all steps (dialing, ring back tone, conversation and release).	☐	☒	☐	
4	**Call to wrong number** (SIP: "404 Not Found") With the SIP phone 141 call a wrong number Check the ring back tone and display	☐	☒	☐	
5	**Call to busy user** **(SIP: "486 Busy Here")** With the SIP phone 141 call 135 take the call and don't hang up. With other SIP phone 145 call 135 which is busy and camp on protection is enabled Check the ring back tone and display	☐	☒	☐	
6	**Call to user in "Out of Service" state** **(SIP: "480 Temporarily not available")** With the SIP phone 141 call the line 135 which is in "Out of Service State" Check the display and ring back tone	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
7	**Call to user in "Do not Disturb" (DND) state (SIP: "480 Temporarily not available")** Dial "*63" on the IP Touch set 134 in order to enable the DND. Wait for acknowledgement ring back tone from OXO. With the SIP phone 141 call the 134. Check ring back tone and display.	☐	☒	☐	Prefix *60 to *69 is configured in the OXO for Call forwarding
8	**Call to local user, immediate forward (CFU) (SIP: "181 Forwarded")(1)** On 134 dial the *61101 (*61 + <target number>) to activate immediate forwarding. Wait for acknowledgement ring back tone from OXO. With 141 call the 134. Check that 101 is ringing. Take the call check audio, the display and hung up. Dial *60 on 134 for forward cancellation.	☐	☒	☐	
9	**Call to local user, forward on no reply (CFNR). (1)** On IP Touch 134 configure with OMC the CFNR using dynamic routing to 102. With 141 call the 134. Check that 134 is ringing but don't take the call. After t1 time out check that 102 is ringing and take the call. Check the audio and display.	☐	☒	☐	When 102 answers, OXO sends a 181 Forwarded and 200 OK - display on 141 is updated
10	**Call to local user, forward on busy (CFB). (1)** On 134 dial the *62101 (*62 + <target number>) to activate the CFB. Wait for acknowledgement ring back tone from OXO. With 134 call 135 and take the call to make it busy. With 141 call 134. Check that 101 is ringing and take the call. Check the audio and display. Dial *60 on 134 for forward cancellation.	☐	☒	☐	
11	**Call to external number (via T2) (Check ring back tone, called party display)** With 141 dial public number of 134 Check that 134 is ringing. Take the call and check audio, display and call release.	☐	☒	☐	
12	**SIP session timer expiration: Check if call is maintained or released after the session timer has expired** With 141 dial 134. Take the call on 134 and never hang up, wait for time out expiration. Check that call is maintained or release.	☒	☐	☐	N/A OmniPCX Office R8.1 doesn't implement SIP session timer for SIP phones

Notes:

(1) For test cases with call to forwarded user: User is forwarded to another local user. Special case of forward to Voice Mail is tested in another section.

8.4 Incoming Calls

8.4.1 Test Objectives

Calls will be generated using the numbers or the name of the SIP user.
SIP terminal will be called in different states: free, busy, out of service, forward.
The states are to be set by the appropriate system prefixes unless otherwise noted.
Points to be checked: tones, voice during the conversation, display (on caller and called party), hang-up phase.

Network calls are made using SIP private trunk established between two OXO's.
OXO prefixes are mandatory for several tests of this section. For more information refer to the appendix C.

8.4.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Local /network call to free SIP terminal Local: with 134 call 141. Check that 141 is ringing and take the call Check ring back tone and called party display. Network: with IP Touch set 240 on OXO 2 call 141. Check that 123 is ringing and take the call. Check ring back tone and called party display.	☐	☒	☐	
2	Local/network call to busy SIP terminal Local: With 102call 141 and take the call to make it busy, don't hang up. With 134 call 141 which is busy Check the ring back tone and display. Network: With 102 call 141 and take the call to make it busy, don't hang up. With 202 call 141 which is busy Check ring back tone and called party display.	☐	☒	☐	
3	Local/network call to unplugged SIP terminal Local: Unplug the 141 SIP set and call it with 134 Check the ring back tone and display Network: Unplug the 143 SIP set and call it with 202 Check the ring back tone and display	☐	☒	☐	call to SIP phone is not routed to its VMU until OXO detects SIP phone is unregistered (after register time out)
4A	**Local/network call to SIP terminal in Do Not Disturb (DND) mode** **By local feature if applicable:** Local: Enable DND on 141 and call it with 134 Check the ring back tone and display Cancel the DND on 141. Network: Enable DND on 141 and call it with 242 Check the ring back tone and display Cancel the DND on 141.	☐	☒	☐	134 displays "released" 242 displays "unobtainable"

Test Case Id	Test Case	N/A	OK	NOK	Comment
4B	**By system feature:** <u>Local</u>: Enable DND on 141 using the prefix *63 prefix. Wait for acknowledgement ring back tone from OXO. With 134 call 141 Check the ring back tone and display Cancel the DND on 398 using prefix *60. <u>Network</u>: Enable DND on 141 using the prefix *63 prefix. Wait for acknowledgement ring back tone from OXO. With 202 call 141 Check the ring back tone and display Cancel the DND on 141 using prefix * 60	☐	☐	☒	DND is activated and working in the call server but OXO returns a wrong SIP msg “500 Internal Server Error” : end user feedback is wrong. DND cancel is working but OXO returns a wrong message “500 Internal Server Error” - end user feedback is wrong.
5A	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to local user:** **By local feature if applicable:** <u>Local</u>: On 141 enable CFU to 135. With 134 call 141. Check that 135 is ringing. Take the call and check audio and display. Disable CFU on 141. <u>Network</u>: On 141 enable CFU to 135. With 240 call 141. Check that 135 is ringing. Take the call and check audio and display. Disable CFU on 141.	☐	☐	☒	Network: display on set 240 shows SIP set calling name even after the forwarding destination rings ; the name is changed only after the call is picked up by the set to which the call is forwarded.
5B	**By system feature :** <u>Local</u>: On 141 enable CFU to 135 using prefix *61135 (*61 + <target number>). Wait for acknowledgement ring back tone from OXO. With 134 call 141. Check that 135 is ringing. Take the call and check audio and display. Disable CFU on 141 using *60 prefix. <u>Network</u>: On 141 enable CFU to 135 using prefix *61135 prefix (*61 + <target number>). Wait for acknowledgement ring back tone from OXO. With 240 call 141. Check that 135 is ringing. Take the call and check audio and display. Disable CFU on 141 using *60 prefix.	☐	☐	☒	CFU is activated in the call server but OXO returns a wrong message “500 Internal Server Error” - end user feedback is wrong (tone)
6A	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to network number:** **By local feature if applicable:** <u>Local</u>: On 141 enable CFU to 202 With 134 call 141. Check that 202 is ringing. Take the call and check audio and display. Disable CFU on 141. <u>Network:</u> On 141 enable CFU to 202. With 240 call 141. Check that 202 is ringing. Take the call and check audio and display.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
	Disable CFU on 141.				
6B	**By system feature :** Local: On 141 enable CFU to 202 using *61202 prefix (*61 + <target number>). Wait for acknowledgement ring back tone from OXO. With 134 call 141. Check that 202 is ringing. Take the call and check audio and display. Disable CFU on 141 using *60 prefix. Network: On 141 enable CFU to 202 using *61202 prefix (*61 + <target number>). Wait for acknowledgement ring back tone from OXO. With 134 call 141. Check that 202 is ringing. Take the call and check audio and display. Disable CFU on 141 using *60 prefix.	☐	☐	☒	CFU is activated in the call server but OXO returns a wrong message “500 Internal Server Error” - end user feedback is wrong (tone)
7A	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to another SIP user** **By local feature if applicable:** Local: On 145 enable CFU to 241 With 141 call 145. Check that 241 is ringing. Take the call and check audio and display. Disable CFU on 145. Network: On 241 enable CFU to 141. With 145 call 241. Check that 141 is ringing. Take the call and check audio and display. Disable CFU on 241.	☐	☒	☐	
7B	**By system feature :** Local: On 145 enable CFU to 241 using *61241 prefix (*61 + <target number>). Wait for acknowledgement ring back tone from OXO. With 141 call 241. Check that 145 is ringing. Take the call and check audio and display. Disable CFU on 145 using *60 prefix. Network: On 241 enable CFU to 141 using *61141 prefix (*61 + <target number>). Wait for acknowledgement ring back tone from OXO. With 145 call 241. Check that 141 is ringing. Take the call and check audio and display. Disable CFU on 241 using *60 prefix.	☐	☒	☐	
8A	**Local call to SIP terminal in “forward on busy” (CFB) state:** **By local feature if applicable** On 141 enable CFB to 135 With 101 and 102 call 141 to make it busy. With 134 call 141 which is busy.	☐	☒	☐	SoundPoint IP 670 is configured with 2 calls per line and is considered as busy on third call

Test Case Id	Test Case	N/A	OK	NOK	Comment
	Check that 135 is ringing Take the call and check audio and display. Disable CFU on 141.				
8B	**By system feature :** On 141 enable CFB to 135 using *62135 prefix (*62 + <target number>). Wait for acknowledgement ring back tone from OXO. With 101 call 141 to make it busy. With 134 call 141 which is busy. Check that 135 is ringing Take the call and check audio and display. Disable CFB on 141 using *60 prefix.	☐	☐	☒	Call forward on busy is not working for SIP phones on OXO
9A	**Local call to SIP terminal in "forward on no reply" (CFNR)** **By local feature if applicable** On 141 enable CFNR to 135 after 2 rings With 101 call 141. Check that 141 is ringing and don't take the call, wait for time out (about 12 seconds). After time out expiration the 135 is ringing, take the call and check audio and display. Disable CFNR on 141.	☐	☒	☐	SoundPoint IP 670 sends to call server 302 Moved Temporarily
9B	**BY SYSTEM FEATURE**	☒	☐	☐	CNFR via prefix not available on OXO (dynamic routing has to be used)
10	External call to SIP terminal. Check that external call back number is shown correctly: With 141 dial public number of 135 Check that 135 is ringing and the external call number is shown correctly Take the call and check audio, display and call release.	☐	☒	☐	
11	Calling Line Identity Restriction (CLIR) : Local call to SIP terminal. Enable CLIR on 134 and call 141 Check that 141 is ringing, take the call and check the display	☐	☒	☐	Display shows "anonymous@anonymous.invalid"
12	Display: Call to free SIP terminal from user with a name containing non-ASCII characters (eg éêêèè). Check caller display. With 135 (extension with a name containing non-ASCII characters) call 141. Check that 141 is ringing and check the name éêêèè is displayed.	☐	☒	☐	
13	Display: Call to free SIP terminal from user with a UTF-8 name containing non-ASCII characters (eg &@@@###). Check caller display. With 135 (extension with a name containing n UTF-8 characters) call 141 Check that 141 is ringing and check the name &@@@###	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
	is displayed.				
14	SIP sets 141 & 145 are part of a sequential hunt group **(1)**. Call to hunt group. Check call/release. With 134call the sequential hunt group number 501 Check that 141 is ringing Take the call and don't hang up. With 135 call the sequential hunt group number 328 Check that 145 is ringing Take the call and don't hang up.	☐	☒	☐	
15	SIP sets 141 & 145 are part of a cyclic hunt group **(2)**. Call to hunt group. Check call/release. With 134 call the cyclic hunt group number 502 Check that 141 is ringing Take the call and hang up. With 134 call the cyclic hunt group number 502 Check that 145 is ringing Take the call and hang up.	☐	☒	☐	
16A	SIP set is configured in a Multiset (secondary). Call to main set and see if Multiset rings. Configure 135 as primary set and 141 as secondary With 134 call 135. Check that 135 and 141 are both ringing. Take the call from 135 and check that 141 stops ringing. Check audio and display. Redo the same test but take the call from 141 and check that 135 stops ringing. Check audio and display.	☐	☒	☐	
16B	SIP set is configured in a Multiset (primary). Call to main set and see if Multiset rings. Configure 141 as primary set and 135 as secondary With 134 call 141. Check that 135 and 141 are both ringing. Take the call from 135 and check that 141 stops ringing. Check audio and display. Don't hang up. With 102 call 141 Check that 141 is ringing. Take the call from 141 check audio and display				

Notes:
(1) Sequential Hunt Group behavior: the endpoint n+1 is ringing **only** if the endpoint n is now in call (busy).

(2) Cyclic Hunt Group behavior: the endpoint n+1 is ringing if previously the endpoint n has been reached (ringing only or in call). The actual state of the n endpoint doesn't matter.

8.5 Features during Conversation

8.5.1 Test Objectives

Features during conversation between local user and SIP user must be checked.
Check that right tones are generated on the SIP phone. A multiline SIP set is mandatory for tests 2, 3, 4 and 8.

OXO prefixes are mandatory for some tests of this section. For more information refer to the appendix C.

8.5.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1A	**Hold and resume with local feature** (if applicable) With 141 call 134 take the call, check audio and display. With 141 put 134 on hold check tones and display on both and resume the call. With 134 put 141 on hold check tones and display on both and resume the call. Keep this call for the next test.	☐	☒	☐	
1B	**Enquiry call to another local user** (if applicable) Distant user is put on hold with local feature With 141 (which is multi-lines) call 135 and take the call 134 will be put on hold when making second call to 135 Put 135 on hold and check tones and display on both. Keep these two calls for the next test.	☐	☒	☐	
1C	**Broker request, toggle back and forth between both lines with local feature** (if applicable) With 141 switch between 134 and 135 lines. Check the tones and display on sets on hold state. Keep these two calls for the next test.	☐	☒	☐	
1D	Release first call. Keep second call. Hang up 134 and only 141 and 135 are in call Check that 135 and 141 are still in call and display.	☐	☒	☐	
2	Repeat the test 1C to 1D but using the call server feature	☒	☐	☐	Hold, enquiry, broker call functionality is not supported within call server for SIP device
3	**Three party conferences initiated from OXO set** With 134 call 141, take the call and don't release it. With 134 call 145, take the call and don't release it too. With 134 start a conference. Check that 134, 141 and 145 are in conference. Check audio and display.	☐	☒	☐	SIP device display is not updated when OXO users initiate the conference

Test Case Id	Test Case	N/A	OK	NOK	Comment
8A	**Three party conferences initiated from SIP set with local feature** (if applicable) With 141 call 134 take the call and don't release it. With 141 call 135, take the call and don't release it too. With 141 start a conference by the local feature Check that 141, 134 and 135 are in conference. Check audio and display.	☐	☒	☐	Display on IP Touch sets doesn't show the second user in the conference initiated on the SIP device
8B	**Three party conferences initiated from SIP set with local feature**	☒	☐	☐	Conference feature is not supported within call server for SIP device
9	**Meet Me conference** With 141 call the Meet me Conference bridge dialing prefix 68 and follow instruction to open the bride. With 145 join the conference bridge by dialing prefix 69 and enter access code. With 134 join the conference bridge by dialing prefix 69 and enter access code. Check that 141, 145 and 134 are in conference.	☐	☒	☐	

8.6 Call Transfer

8.6.1 Test Objectives

During the consultation call step, the transfer service can be requested and must be tested.
Several transfer services exist: Unattended Transfer, Semi-Attended Transfer and Attended Transfer.
Audio, tones and display must be checked.

We use the following scenario, terminology and notation:

There are three actors in a given transfer event:

- A – *Transferee*: the party being transferred to the Transfer Target.
- B – *Transferor*: the party doing the transfer.
- C – *Transfer Target*: the new party being introduced into a call with the Transferee.

There are three sorts of transfers in the SIP world:

- ***Unattended Transfer*** or *Basic Transfer or Blind Transfer* : The Transferor provides the Transfer Target's contact to the Transferee. The Transferee attempts to establish a session using that contact and reports the results of that attempt to the Transferor.

- ***Semi-Attended Transfer*** or *Early Attended Transfer* or *Transfer on ringing*:

1. A (Transferee) calls B (Transferor).
2. B (Transferor) calls C (Transfer Target). A is on hold during this phase. C is in ringing state (does not pick up the call).

3. B executes the transfer. B drops out of the communication. A is now in contact with C, in ringing state. When C picks up the call it is in conversation with A.

- ***Attended Transfer*** or *Consultative Transfer* or *Transfer in conversation*:
  1. A (Transferee) calls B (Transferor).
  2. B (Transferor) calls C (Transfer Target). A is on hold during this phase. C picks up the call and goes in conversation with B.
  3. B executes the transfer. B drops out of the communication. A is now in conversation with C.

Note: Unattended and Semi Attended are not supported for SIP phones on OmniPCX Office.

8.6.2 Test Results Attended Transfer

In the below table, *SIP* means the partner SIP device, *OXO* means a proprietary OXO (IPTouch/UA/Z) set, *Ext. Call* means an External Call, ISDN for example.

Test	Action			Result	Comment
	A Transferee	B Transferor	C Transfer Target		
1	OXO 134	SIP 141	OXO 135	OK	
2	Ext Call	SIP 141	OXO 134	OK	
3	Ext Call	SIP 141	Ext Call	OK	
4	SIP 145	SIP 141	SIP 241	OK	
5	SIP 141	OXO 134	OXO 135	OK	
6	Ext Call	OXO 134	SIP 141	OK	
7	SIP 141	OXO 134	SIP 145	OK	

8.7 Attendant

8.7.1 Test Objectives

An attendant console is defined on the system. Call going to and coming from the attendant console are tested.

8.7.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**SIP set Call to attendant (using attendant call prefix "9")** From 141 dial "9" Check audio and display	☐	☒	☐	Display on 141 shows internal number/name of Attendant extension
2	**2nd incoming call while in conversation with attendant.** While 141 is in conversation with the attendant, from 134 call 141. Answer the call and check audio and display.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
3	**SIP set call to attendant (using attendant call prefix “9”), attendant transfers to OXO set, unattended.** 141 is in conversation with the attendant. Attendant transfer unattended to 134 Answer the call and check audio and display.	☐	☒	☐	
4	**Call to attendant (using attendant call prefix “9”), attendant transfers to OXO set, attended.** 141 is in conversation with the attendant. Attendant transfer attended to 134 Check audio and display.	☐	☒	☐	
5	**OXO set calls to attendant (using attendant call prefix “9”), attendant transfers to SIP set, unattended.** 134 is in conversation with the attendant. Attendant transfer attended to 141. Check audio and display.	☐	☒	☐	
6	**OXO set calls to attendant (using attendant call prefix “9”), attendant transfers to SIP set, attended.** 134 is in conversation with the attendant. Attendant transfer attended to 141. Check audio and display.	☐	☒	☐	

8.8 Voice Mail

8.8.1 Test Objectives

Voice Mail notification, consultation and password modification must be checked.
MWI (Message Waiting Indication) has to be checked.

The default Voice Mail number is 500 and this service is enabled on SIP sets 141 and 145 and IP Touch 134.
For these tests, DTMF sending (RFC 2833) has to be validated in order to use Voice Mail menu.

8.8.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Password configuration. With 141 call the Voice Mail at 500 and follow the Voice guide in order to modify the default password. When modification is accepted hang-up. Recall the voice mail and try to log with a wrong password. Check the rejection. Recall the voice mail and try to log with the right password. Check the service access.	☐	☒	☐	
2	Message display activation, MWI (1): With 145 call the Voice Mail at 500 Follow the instructions in order to send a voice message in 141 boxes. Check that the MWI on 141 is activated.	☐	☒	☐	There is no count of messages on the display. Notification is only for messages.
3	Message consultation. With 141 call the Voice Mail at 500. Follow the instructions in order to listen your voice message leaved during the previous test. Check that your can listen it and delete. Check that MWI display is disabled on 141 after message cancellation.	☐	☒	☐	
4	OXO set call to a SIP user forwarded to Voice Mail Forward the 141 to Voice Mail by dialing *61500 (*61 prefix + <Voice Mail number>). With 134 call 141 and check that you are immediately forwarded to Voice Mail. Check that you can leave a message On 141 disable forwarding to VMU with prefix *60.	☐	☒	☐	
5	SIP call to a OXO user forwarded to Voice Mail. Forward the 134 to Voice Mail by dialing *61500 (*61 prefix + <Voice Mail number>). With 141 call 134 and check that you are immediately forwarded to Voice Mail. Check that you can leave a message	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
	On 134 disable forwarding to VMU with prefix *60.				

Notes:
(1) On SIP sets, in order to enable the MWI feature, you have to configure the Voice Mail number.

8.9 Defence

8.9.1 Test objectives

Check how the SIP set will react in case of a OXO reboot, Ethernet link failure.

8.9.2 Test procedure

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**OXO Reboot** Establish an incoming ISDN call with 141. Reboot the OXO. When the OXO is up again, re-establish an incoming ISDN call with 141 and check the audio.	☐	☒	☐	
2	**Ethernet link failure** Establish an incoming ISDN call with 141. Disconnect the Ethernet link of 141. Check that the incoming call is presented to the attendant. Reconnect the Ethernet link of 141. Re-establish an incoming ISDN call with 141 and check the audio.	☐	☒	☐	ISDN call is rerouted to attendant after register expire time (120s by default).

9 Appendix A: AAPP member's Application description

SoundPoint IP 670:

Configure phone IP address using MMI interface

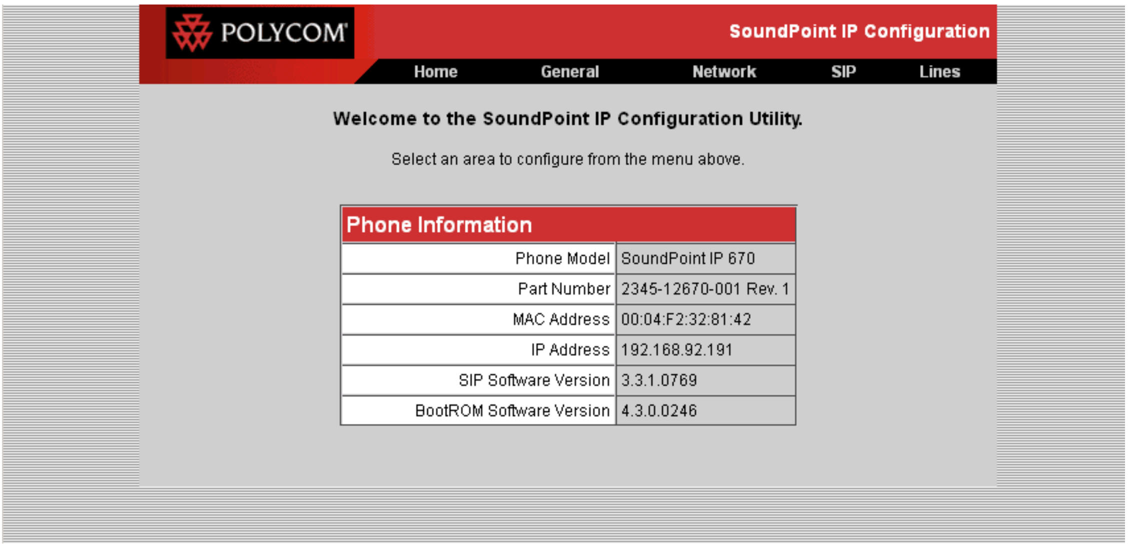
Enter the phone IP address

1. During startup process select Setup, enter password (default: 456)
2. Disable DHCP client
3. Configure phone's IP address / subnet mask / IP gateway (OXO)
4. Save and reboot the device
5. After restart network settings and other parameters can modified in MENU – 3. Settings – Advanced – Admin Settings

10 Appendix B: Configuration requirements of the AAPP member's application

Access to Admin Homepage (web interface)

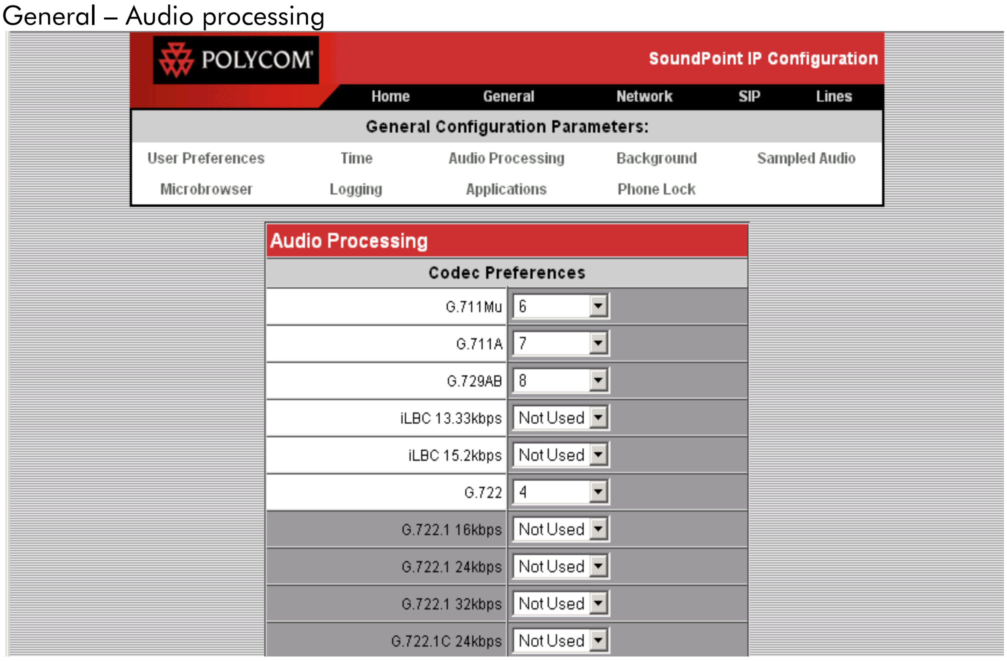
1. Open a web browser
2. Enter the SoundPoint IP 670 IP address in the address bar ex: http://192.168.92.191
3. Enter the administrator login "Polycom"
4. Enter the administrator password (default value: 456)
    4. You will access the Homepage of the SoundPoint IP 670
    5.



Note : static IP parameters are configured manually on the device (MMI)

CODEC configuration

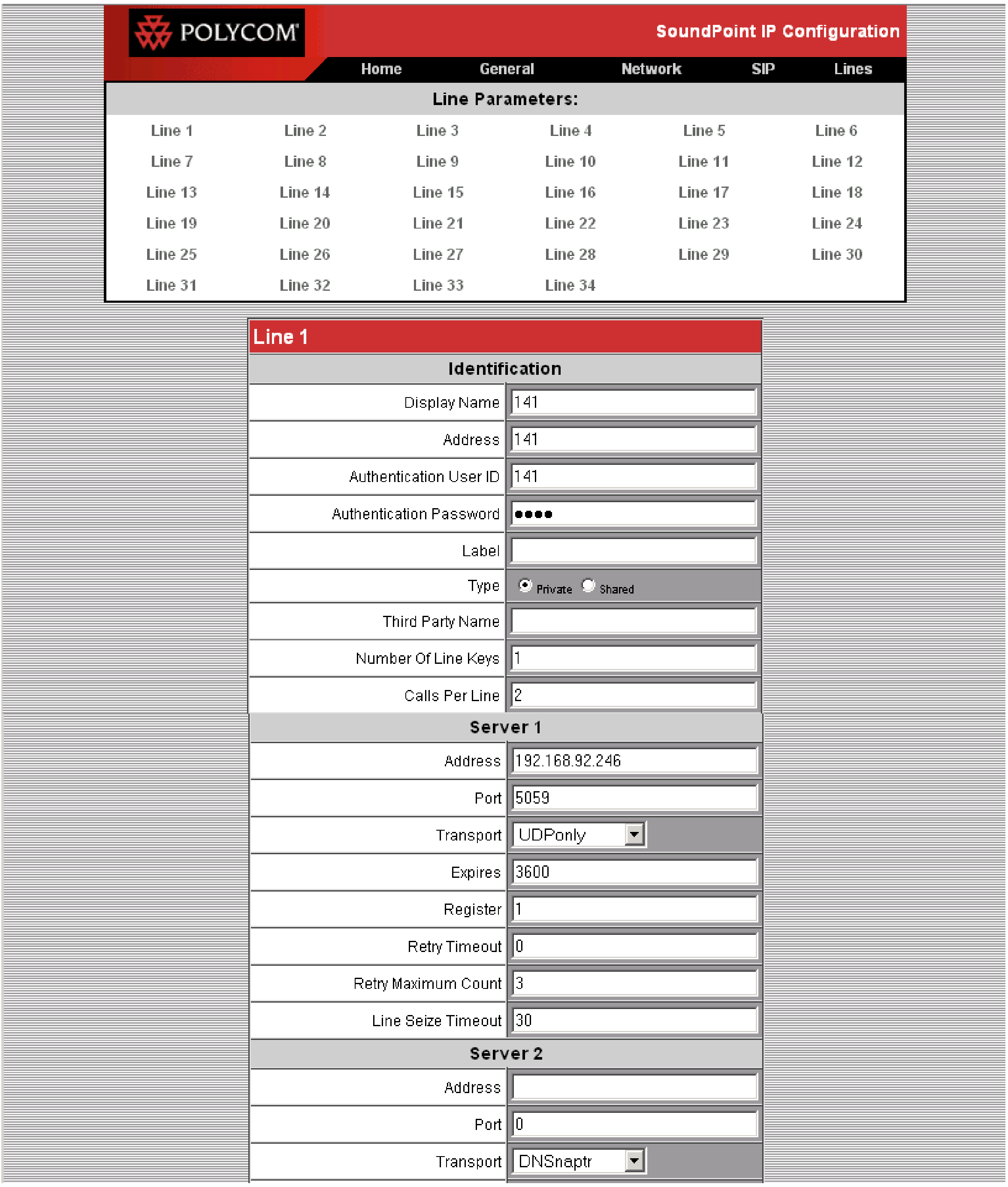
General – Audio processing



Enable the required CODEC and keep the priority of the CODEC as required (unless otherwise stated).

SIP Registrar / Authentication

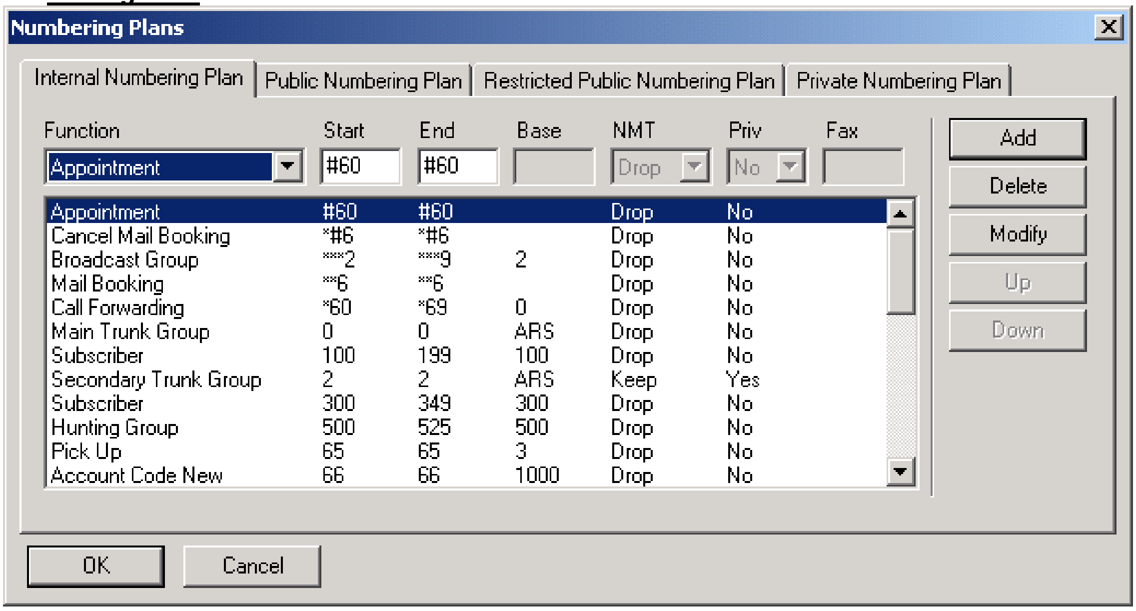
Lines – Line 1



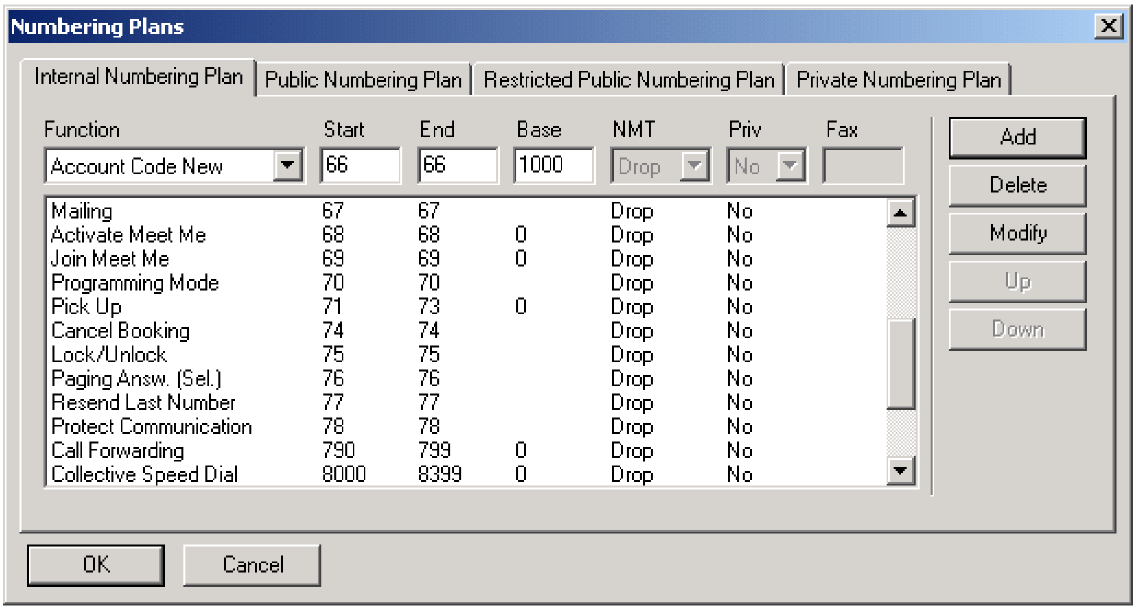
11 Appendix C: Alcatel-Lucent Communication Platform: configuration requirements

OXO Configuration overview

1. Dialing Plan

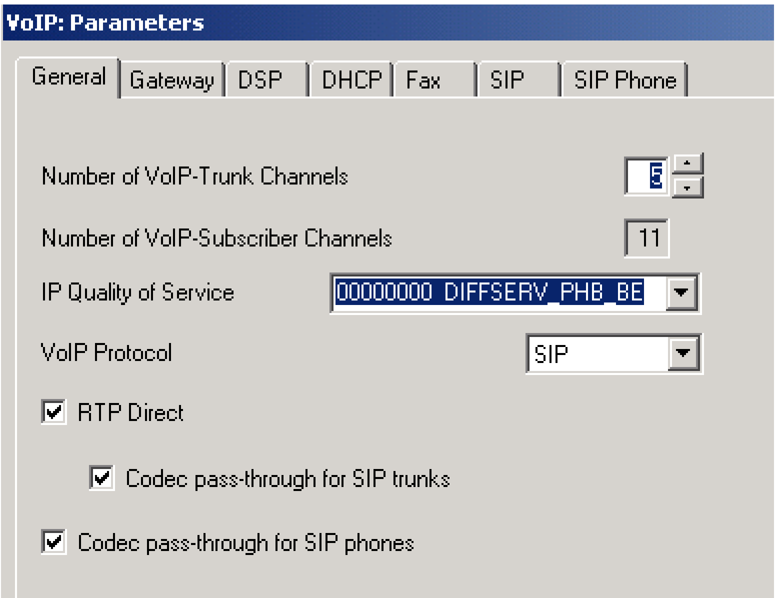


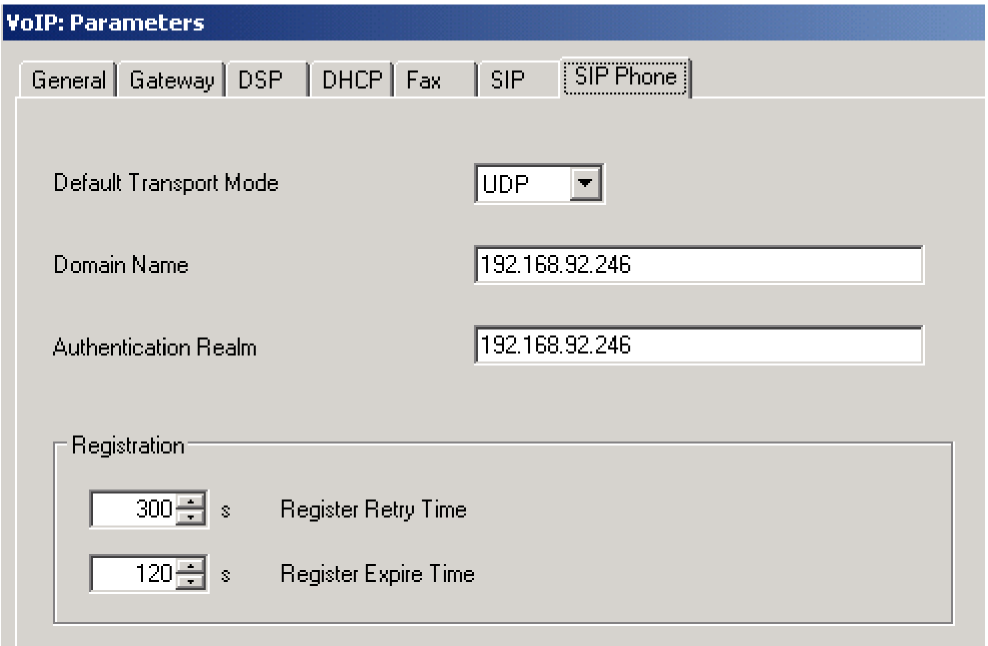
Function	Start	End	Base	NMT	Priv
Appointment	#60	#60		Drop	No
Cancel Mail Booking	*#6	*#6		Drop	No
Broadcast Group	***2	***9	2	Drop	No
Mail Booking	**6	**6		Drop	No
Call Forwarding	*60	*69	0	Drop	No
Main Trunk Group	0	0	ARS	Drop	No
Subscriber	100	199	100	Drop	No
Secondary Trunk Group	2	2	ARS	Keep	Yes
Subscriber	300	349	300	Drop	No
Hunting Group	500	525	500	Drop	No
Pick Up	65	65	3	Drop	No
Account Code New	66	66	1000	Drop	No



Function	Start	End	Base	NMT	Priv
Mailing	67	67		Drop	No
Activate Meet Me	68	68	0	Drop	No
Join Meet Me	69	69	0	Drop	No
Programming Mode	70	70		Drop	No
Pick Up	71	73	0	Drop	No
Cancel Booking	74	74		Drop	No
Lock/Unlock	75	75		Drop	No
Paging Answ. (Sel.)	76	76		Drop	No
Resend Last Number	77	77		Drop	No
Protect Communication	78	78		Drop	No
Call Forwarding	790	799	0	Drop	No
Collective Speed Dial	8000	8399	0	Drop	No

2. VoIP Configuration



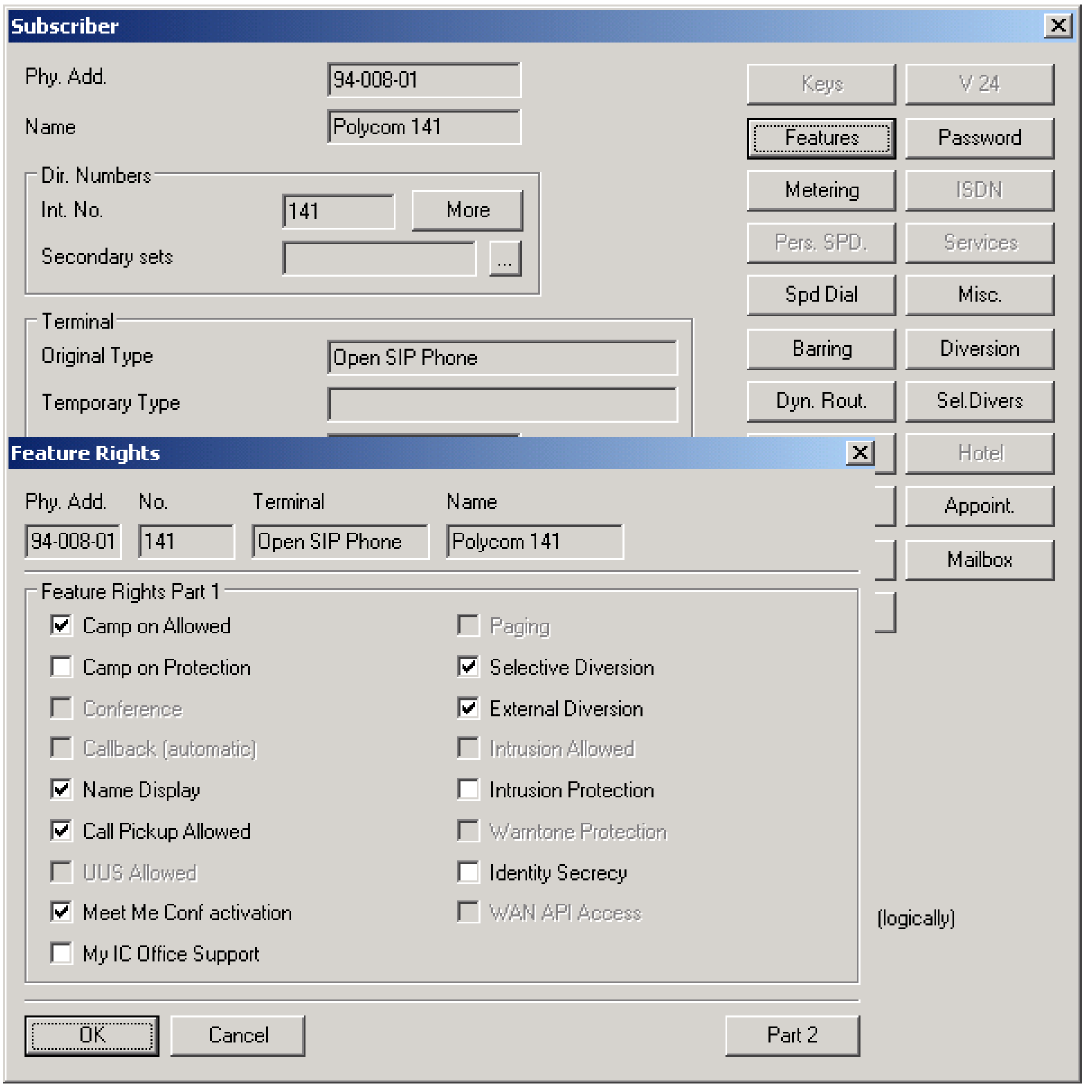


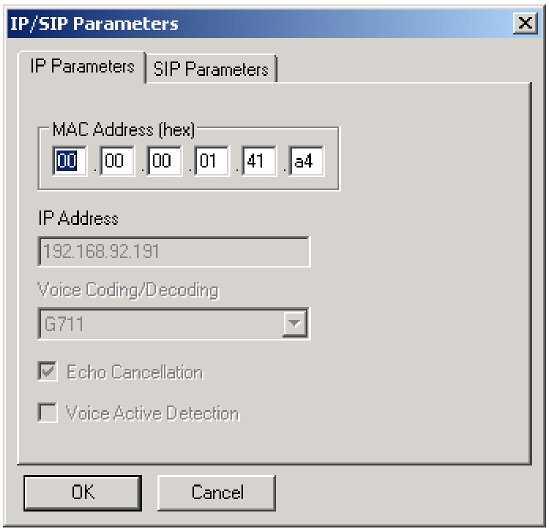
3. Network Call Configuration:

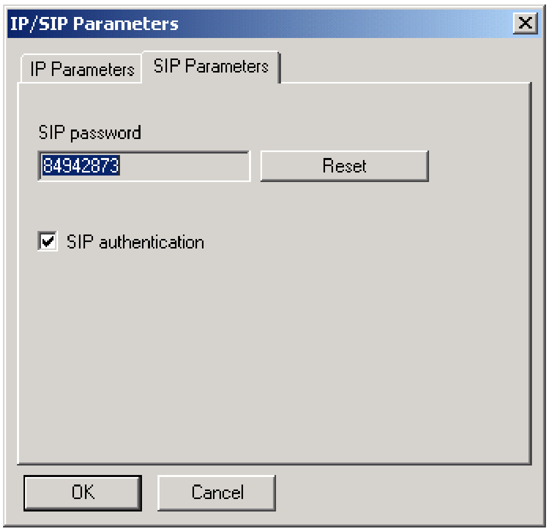
Automatic Routing: Prefixes

Activation	Network	Prefix	Ranges	Substitute	TrGpList	Called(ISVPN/H450)	User comment	Destination	IP Type	IP Address	Hc
Yes	pub				1	het		Not IP			
Yes	priv	2		2	2	het	OXO2	SIP Gateway	Static	192.168...	

4. SIP Set Configuration







12 Appendix D: AAPP Member's escalation process

Polycom Global Services (PGS) provides support to Polycom VoIP certified resellers or Polycom SoundStation IP Certified Resellers only (referred to below as "Certified Resellers" or "CR"). Support is limited to units under warranty or units under a valid support contract. If product is not under warranty or under a current support contract, there is an option to obtain technical support through Pay-Per-Incident services.

SoundPoint IP 670 escalation support process for Certified Resellers:

- End customer contacts Certified Reseller (CR) for service

- CR opens ticket and determines that there is a hardware or software issue with the Polycom IP phone and attempts to solve issue.

- If needed CR opens case with PGS by calling Polycom's Technical Assistance Center: call center at: 1-888-248-4143, option #2, option #1

- Before calling Polycom, Certified Resellers will need:

    * CPC code (certified Partner code)
    * Serial Number (MAC address) of the phones they are calling about

- Data required to effectively troubleshoot issues:
    Clear description of the issue
        * Reboot, lockup, no/poor audio, echo etc
    Number of phones/users/sites affected
    Frequency of occurrence / Ability to reproduce
    Software version in use / other versions tested
    Log files
    Ethernet packet capture taken at the phone

Configuration files

13 Appendix E: AAPP program

13.1 Alcatel-Lucent Application Partner Program (AAPP)

The Application Partner Program is designed to support companies that develop communication applications for the enterprise market, based on Alcatel-Lucent's product family. The program provides tools and support for developing, verifying and promoting compliant third-party applications that complement Alcatel-Lucent's product family. Alcatel-Lucent facilitates market access for compliant applications.

The Alcatel-Lucent Application Partner Program (AAPP) has two main objectives:

- **Provide easy interfacing for Alcatel-Lucent communication products**: Alcatel-Lucent's communication products for the enterprise market include infrastructure elements, platforms and software suites. To ensure easy integration, the AAPP provides a full array of standards-based application programming interfaces and fully-documented proprietary interfaces. Together, these enable third-party applications to benefit fully from the potential of Alcatel-Lucent products.

- **Test and verify a comprehensive range of third-party applications**: to ensure proper inter-working, Alcatel-Lucent tests and verifies selected third-party applications that complement its portfolio. Successful candidates, which are labelled Alcatel-Lucent Compliant Application, come from every area of voice and data communications.

The Alcatel-Lucent Application Partner Program covers a wide array of third-party applications/products designed for voice-centric and data-centric networks in the enterprise market, including terminals, communication applications, mobility, management, security, etc.

Web site

The Application Partner Portal is a website dedicated to the AAPP members and potential candidates. It can be accessed at this URL:
http://applicationpartner.alcatel-lucent.com

13.2 Alcatel-Lucent.com

You can access the Alcatel-Lucent website at this URL: http://www.Alcatel-Lucent.com/

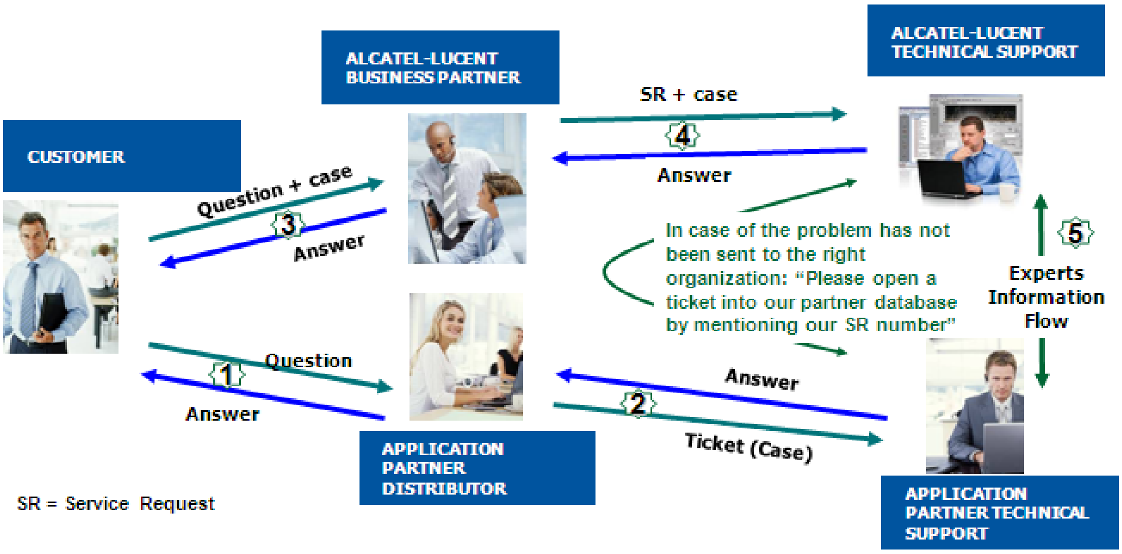
14 Appendix F: AAPP Escalation process

14.1 Introduction

The purpose of this appendix is to define the escalation process to be applied by the Alcatel-Lucent Business Partners when facing a problem with the solution certified in this document.

The principle is that Alcatel-Lucent Technical Support will be subject to the existence of a valid InterWorking Report within the limits defined in the chapter “Limits of the Technical support”.

In case technical support is granted, Alcatel-Lucent and the Application Partner are engaged as following:



(*) The Application Partner Business Partner can be a Third-Party company or the Alcatel-Lucent Business Partner itself

14.2 Escalation in case of a valid Inter-Working Report

The InterWorking Report describes the test cases which have been performed, the conditions of the testing and the observed limitations.

This defines the scope of what has been certified.

If the issue is in the scope of the IWR, both parties, Alcatel-Lucent and the Application Partner, are engaged:

Case 1: the responsibility can be established 100% on Alcatel-Lucent side.
In that case, the problem must be escalated by the ALU Business Partner to the Alcatel-Lucent Support Center using the standard process: open a ticket (eService Request –eSR)

Case 2: the responsibility can be established 100% on Application Partner side.
In that case, the problem must be escalated directly to the Application Partner by opening a ticket through the Partner Hotline. In general, the process to be applied for the Application Partner is described in the IWR.

Case 3: the responsibility can not be established.
In that case the following process applies:

- The Application Partner shall be contacted first by the Business Partner (responsible for the application, see figure in previous page) for an analysis of the problem.

- The Alcatel-Lucent Business Partner will escalate the problem to the Alcatel-Lucent Support Center only if the Application Partner has demonstrated with traces a problem on the Alcatel-Lucent side or if the Application Partner (not the Business Partner) needs the involvement of Alcatel-Lucent.

In that case**,** the Alcatel-Lucent Business Partner must provide the reference of the Case Number on the Application Partner side. The Application Partner must provide to Alcatel-Lucent the results of its investigations, traces, etc, related to this Case Number.

Alcatel-Lucent reserves the right to close the case opened on his side if the investigations made on the Application Partner side are insufficient or do no exist.

Note: Known problems or remarks mentioned in the IWR will not be taken into account.

For any issue reported by a Business Partner outside the scope of the IWR, Alcatel-Lucent offers the "On Demand Diagnostic" service where Alcatel-Lucent will provide 8 hours assistance against payment.

IMPORTANT NOTE 1: The possibility to configure the Alcatel-Lucent PBX with ACTIS quotation tool in order to interwork with an external application is not
the guarantee of the availability and the support of the solution. The reference remains the existence of a valid InterWorking Report.

Please check the availability of the Inter-Working Report on the AAPP (URL: https://private.applicationpartner.alcatel-lucent.com) or Enterprise Business Portal (Url: Enterprise Business Portal) web sites.

IMPORTANT NOTE 2: Involvement of the Alcatel-Lucent Business Partner is mandatory, the access to the Alcatel-Lucent platform (remote access, login/password) being the Business Partner responsibility.

14.3 Escalation in all other cases

These cases can cover following situations:

1. An InterWorking Report exist but is not valid (see Chap 2 "Validity of an Interworking Report")

2. The 3rd party company is referenced as AAPP participant but there is no official InterWorking Report (no IWR published on the Enterprise Business Portal for Business Partners or on the Alcatel-Lucent Application Partner web site) ,

3. The 3rd party company is NOT referenced as AAPP participant

In all these cases, Alcatel-Lucent offers the "On Demand Diagnostic" service where Alcatel-Lucent will provide 8 hours assistance against payment.

14.4 Technical support access

The Alcatel-Lucent **Support Center** is open 24 hours a day; 7 days a week:

- e-Support from the Application Partner Web site (if registered Alcatel-Lucent Application Partner): http://applicationpartner.alcatel-lucent.com
- e-Support from the Alcatel-Lucent Business Partners Web site (if registered Alcatel-Lucent Business Partners): https://businessportal.alcatel-lucent.com click under "Let us help you" the *eService Request* link
- e-mail: Ebg_Global_Supportcenter@alcatel-lucent.com
- Fax number: +33(0)3 69 20 85 85
- Telephone numbers:

Alcatel-Lucent Business Partners Support Center for countries:

Country	**Supported language**	**Toll free number**
France	French	+800-00200100
Belgium		
Luxembourg		
Germany	German	
Austria		
Switzerland		
United Kingdom	English	
Italy		
Australia		
Denmark		
Ireland		
Netherlands		
South Africa		
Norway		
Poland		
Sweden		
Czech Republic		
Estonia		
Finland		
Greece		
Slovakia		
Portugal		
Spain	Spanish	

For other countries:

English answer :	+ 1 650 385 2193
French answer :	+ 1 650 385 2196
German answer :	+ 1 650 385 2197
Spanish answer :	+ 1 650 385 2198

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