Alcatel Lucent Application Partner Program
Inter-Working Report

Partner: **Yealink**
Application type: **SIP phone**
Application name: **Yealink Models T28P, T26P, T22P, T20P**
Alcatel-Lucent Platform: **OmniPCX Office™**

The product and release listed have been tested with the Alcatel-Lucent Communication Platform and the release specified hereinafter. The tests concern only the inter-working between the AAPP member's product and the Alcatel-Lucent Communication Platform. The inter-working report is valid until the AAPP member's product issues a new major release of such product (incorporating new features or functionality), or until Alcatel-Lucent issues a new major release of such Alcatel-Lucent product (incorporating new features or functionalities), whichever first occurs.

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Certification overview

Date of the certification	May 2013

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AAPP member representative	Pablo Wang

Alcatel-Lucent Communication Platform	OmiPCX Office
Alcatel-Lucent Communication Platform Release	R910/021.001
AAPP member application version	T20P 9.70.0.140 T22P 7.70.0.140 T26P 6.70.0.140 T28P 2.70.0.140
Application Category	Terminals

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History:

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Test results

☑ Passed ☐ Refused ☐ Postponed

☐ Passed with restrictions

Refer to the section 6 for a summary of the test results.

IWR validity extension

None

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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.

Note: This interworking report does not cover configuration/management and/or mass provisioning of the SIP phone. For any questions related to these topics, please contact Yealink.

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: *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 Version

Application commercial name: YealinK SIP phone

Application version:

T20P	9.70.0.140
T22P	7.70.0.140
T26P	6.70.0.140
T28P	2.70.0.140

Interface type : SIP/Ethernet

Interface version (if relevant):

Brief application description:

The Yealink SIP-T28P represents the next generation VoIP phone designed for business users who need rich telephony features, a friendly user-interface and superb voice quality. Equipped with the TI TITAN chipset, it offers high-definition voice quality through a TI voice engine, HD handset, HD speaker and HD codec (G.722). The large, high-resolution graphical display, combined with up to 48 keys, guarantees an excellent user experience in terms of configuration options, making calls and to access the express XML browse. To ensure that your audio data remains confidential, the Yealink SIP-T28P also supports security standards TLS, SRTP, HTTPS, 802.1x, Open VPN and AES encryption .This guard against electronic eavesdropping and data theft.

Type of application/product:

T20 P

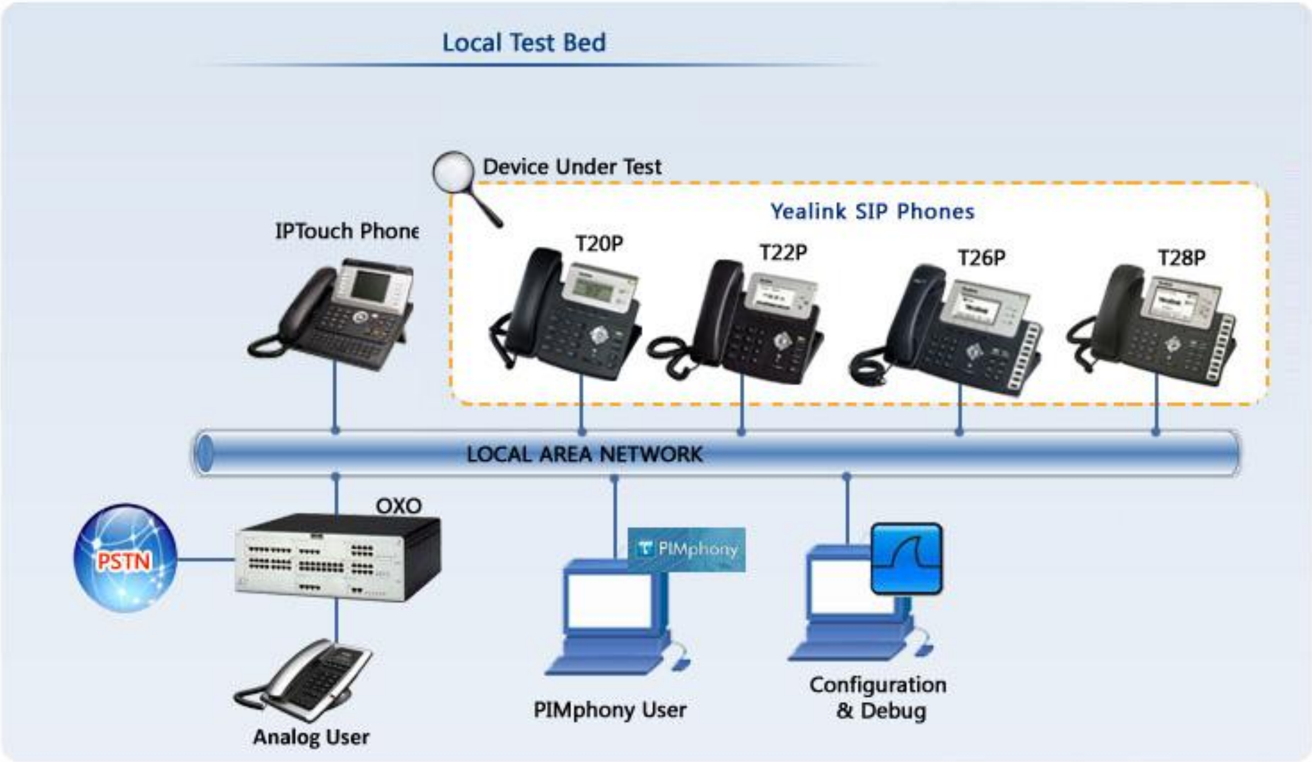
T22P

T26 P

T28P

5 Test Environment

Figure 1 **Test Environment**



5. 1 Hardware configuration

Alcatel-Lucent Communication Platform:

- OmniPCX Office Rack
- PowerCPU
- Release: R910/021.001
- OMC: R910_14.1b

Setup Details:

Setup Information OXO 1	
OXO 1 IP address	10.130.158.48
Domain name	oxoone.testandvalidate.com
Voicemail No	215
Attendant No	0
OXO Extension Details used for test	
IP Touch numbers	226, 227,228
Yealink SIP phones	235 , 236 , 237 , 238
UA Set No	200
Setup Information OXO 2	
Network OXO address	10.130.158.45
Network OXO Domain name	oxotwo.testandvalidate.com
Network OXO Extension Details used for test	
IP Touch numbers	102,103 & 104
UA Set No	101

Note:

1) The Two OXO systems are connected via private SIP Trunk.
2) For some tests we will change the set type from IP Touch to UA set or Analog set.

5. 2 Software configuration

- **Alcatel-Lucent Communication Platform**: OmniPCX Office R910/021.001

- **Partner Application**: Yealink SIP phones
  T20P 9.70.0.140
  T22P 7.70.0.140
  T26P 6.70.0.140
  T28P 2.70.0.140

Note: Yealink Phones are 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		
OXO Internal DHCP server	N/A	DHCP registration is not supported for SIP phones
External DHCP server	OK	
SIP authentication	OK	
Voice over IP and RTP codec support	OK	
Outgoing Call	OK	
Incoming Call	OK	
Features During Conversation	OK	
Call Transfer	OK	Only attended transfer is supported.
Conference	OK	Only Conference with local feature is supported.
Attendant	OK	
Voice mail interaction and indication	OK	

6. 2 Summary of problems

None

6. 3 Summary of limitations

- DHCP mode from OXO is restricted to ALU IP Phones.
- In Conference we are unable to see the other user information other than the one who initiated the conference.
- Call server feature activation (eg CFU/CFB) is not displayed on the Yealink phones.
- No count of new messages (voice mails) available on the display of sip phones. This is an OXO limitation.
- Semi attended and blind Transfer are not supported in OXO.

6. 4 Notes, remarks

- Yealink phones are 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

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).

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**SIP sets** Configure your SIP sets MCDU number on the OXO as 235, 236 & 237 to register with the OXO IP address Check the registration on your sets and the display Note that authentication is disabled for these users, the password doesn't matter.	☐	☒	☐	
2	**SIP set registration to OXO in static IP addressing** For this test we will try to register the SIP phone with authentication enabled. SIP phones 235, 236 & 237 are configured with a static IP address of OXO. Check the phone registration and display. Redo the same test on one IP phone with a wrong password and check that the phone is rejected.	☐	☒	☐	
3A	**DHCP registration (with OXO internal DHCP server)** Check whether the SIP set is able to get IP address from an OXO internal DHCP server.	☒	☐	☐	Restricted to ALE IP phones
3B	**DHCP registration (Use external DHCP server)** Check whether the SIP set is able to get IP address from an external DHCP server.	☐	☒	☐	
4	**NTP registration** The SIP phone 235 is 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 235 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 236 to use the protocol SIP over UDP and other TCP In the two cases, check the registration and basic calls.	☐	☒	☐	

8. 2 Audio codec negotiations/ VAD / Framing

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

Phone configuration: configure YEALINK to use G.722, G.711 A-law, G.711 mu-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).

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Select G711 A-law as 1st codec in YEALINK Call from SIP 235 to IP Touch **226** Check that the call is established in G711 A-law. Check audio quality Call from IP Touch 226 to SIP 236 Check that the call is established in G711 A-law. Check audio quality	☐	☒	☐	
2	Select G729 as 1st codec in YEALINK Call from SIP 235 to IP Touch 226 Check that the call is established in G729 Check audio quality Call from IP Touch 226 to SIP 236 Check that the call is established in G729 Check audio quality	☐	☒	☐	
3	Select G723 as 1st codec in YEALINK Check that the call is established in G723 Check audio quality Call from IP Touch 226 to SIP 235 Check that the call is established in G723 Check audio quality	☐	☒	☐	
4	**Configure** 235 **to use VAD (to enabled in the client end)** **Configure IP Touch** 323 **NOT to use VAD** Call from SIP 235 to IP Touch **226** Check that the call is established in G711 A-law. Check audio quality **Configure SIP** 235 **to use VAD (to enabled in the client end)** **Configure IP Touch** 235 **to use VAD** Redo the same tests. **Configure SIP** 235 **NOT to use VAD** **Configure IP Touch** 226 **to use VAD** Redo the same tests	☐	☒	☐	
5	**In OXO enable codec pass through for SIP phones.** **Call from SIP** 235 **to SIP** 226 Check that the call is established using G.729 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. G729 is preferred codec in YEALINK** **Call from SIP** 235 **to Network SIP** 122 Check that the call is established using direct RTP in G729. 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. G729 is preferred codec in YEALINK** **Call from SIP** 235 **to Network SIP** 122 Check that the call is established in G711. 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 G729_30** **Default codec in sip phones should be set to G711** **Call from SIP** 235 **to Network SIP** 122 Check that the call is established in G729. Check audio quality	☐	☒	☐	

8. 3 Outgoing Calls

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

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

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Call to a local user** With SIP Phone 235 call the IP Touch 226. Check that 226 is ringing. Take the call and check ring back tone audio and display.	☐	☒	☐	
2	**Call to local user with no answer** With SIP Phone 235 call the IP Touch 226. And never take the call. Check time out (if any) and display. Note that 322 don't have a Voice Mail	☐	☒	☐	
3	**Call to another SIP set** With the SIP phone 235 call the other SIP Phone 226	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
	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 235 call a wrong number Check the ring back tone and display	☐	☒	☐	
5	**Call to busy user** **(SIP: "486 Busy Here")** With the SIP phone 235 call IP Touch 226, take the call and don't hang up. With other SIP phone 235 call 225 which is busy Check the ring back tone and display	☐	☒	☐	
6	**Call to user in "Out of Service" state** **(SIP: "480 Temporarily Unavailable")** With the SIP phone 235 call the IP Touch 226 which is in "Out of Service State" Check the display and ring back tone	☐	☒	☐	Call goes to voicemail when not reachable.
7	**Call to user in "Do not Disturb" (DND) state** **(SIP: "480 Temporarily not available")** Dial "*63" on the IP Touch 226 in order to enable the DND. Wait for acknowledgement ring back tone from OXO. With the SIP phone 235 call 226. Check ring back tone and display. Redial *63 on 322 to cancel the DND	☐	☒	☐	
8	**Call to local user, immediate forward (CFU).** **(SIP: "181 Forwarded")(1)** On IP Touch 226 dial the *612268(*61 + 228) to activate the CFU. Wait for acknowledgement ring back tone from OXO. With the SIP phone 235 call the 226. Check that 228 is ringing and the display. Take the call check audio and hung up. Dial *60 on 226 for forward cancellation.	☐	☒	☐	
9	**Call to local user, forward on no reply (CFNR). (1)** On IP Touch 226 configure with OMC the CFNR using dynamic routing to 228. With 235 call the 226. Check that 228 is ringing but don't take the call and wait the time out (about 30 sec). Time out is defined in 322 dynamic routing of Timer 1. After time out check that 226 is ringing and take the call. Check the audio and display.	☐	☒	☐	
10	**Call to local user, forward on busy (CFB). (1)** On IP Touch 226 dial the *62228 (*62+<target MCDU number>) to activate the CFB. Wait for acknowledgement ring back tone from OXO. With SIP phone 235 call 226 and take the call to make it busy.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
	With other SIP phone 236 Call 226. Check that 228 is ringing and take the call. Check the audio and display. Dial *60 on 226 for forward cancellation.				
11	**Call to external number** **(Check ring back tone, called party display)** With SIP set 235 dial 9 (9 prefix +external number) 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 SIP set 235 call IP Touch 226. Take the call on 226 and never hang up, wait for time out expiration. Check that call is maintained or release.	☐	☒	☐	

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

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

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Local /network call to free SIP terminal** Local: with IP Touch 226 call SIP set 235. Check that 235 is ringing and take the call Check ring back tone and called party display. Network: with IP Touch 226 call SIP set 123 on another Node. Check that 123 is ringing and take the call. Check ring back tone and called party display.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
2	**Local/network call to busy SIP terminal** Local: With SIP set 235 call other SIP set 236 and take the call to make it busy, don't hang up. With IP Touch 226 call 234 which is busy Check the ring back tone and display. Network: With SIP set 235 call SIP set 123 and take the call to make it busy, don't hang up. With 226 call 123 which is busy Check ring back tone and called party display.	☐	☒	☐	
3	**Local/network call to unplugged SIP terminal** Local: Unplug the 235 SIP set and call it with IP Touch 226. Check the ring back tone and display Network: Unplug the SIP set 123 and call it with 226 Check the ring back tone and display	☐	☒	☐	Call goes to voicemail
4A	**Local/network call to SIP terminal in Do Not Disturb (DND) mode** **By local feature if applicable:** Local: Enable DND on SIP set 235 and call it with IP Touch 226 Check the ring back tone and display Cancel the DND on 235. Network: Enable DND on SIP set 123 and call it with IP Touch 226 Check the ring back tone and display Cancel the DND on 123.	☐	☒	☐	
4B	**By system feature** Local: Enable DND on SIP set 235 using the *63 prefix. Wait for acknowledgement ring back tone from OXO. With IP Touch 226 call 235 Check the ring back tone and display Cancel the DND on 235 using *63 prefix. Network: Enable DND on SIP set 123 using the *63 prefix. Wait for acknowledgement ring back tone from OXO. With IP Touch 226 call 123 Check the ring back tone and display Cancel the DND on 123 using * 60 prefix.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
5A	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to local user:** **By local feature if applicable:** Local: On SIP set 235 enable CFU to IP Touch 226 With SIP set 236 call 235. Check that 226 is ringing. Take the call and check audio and display. Disable CFU on 235. Network: On SIP set 123 enable CFU to IP Touch 126. With SIP set 235 call 123. Check that 126 is ringing. Take the call and check audio and display. Disable CFU on 123.	☐	☒	☐	
5B	**By system feature:** Local: On SIP set 236 enable CFU to IP Touch 226 using *61226 prefix (*61 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP set 235 call 236. Check that 226 is ringing. Take the call and check audio and display. Disable CFU on 236 using *60 prefix. Network: On SIP Set 123 enable CFU to IP Touch 102 using *61122 prefix (*61 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP Set 235 call 123. Check that 102 is ringing. Take the call and check audio and display. Disable CFU on 123 using *60 prefix.	☐	☒	☐	
6A	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to network number:** **By local feature if applicable:** Local: On SIP Set 235 enable CFU to SIP Set 122.With SIP set 236 call 235. Check that 122 is ringing. Take the call and check audio and display. Disable CFU on 235. Network: On SIP Set 235 enable CFU to IP Touch 102. With SIP Set 123 call 235. Check that 102 is ringing. Take the call and check audio and display. Disable CFU on 235.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
6B	**By system feature:** <u>Local</u>: On SIP Set 235 enable CFU to SIP Set 122 using *61122 prefix (*61 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP set 236 call 235. Check that 122 is ringing. Take the call and check audio and display. Disable CFU on 235 using *60 prefix. <u>Network</u>: On SIP Set 235 enable CFU to IP Touch 102 using *61102 prefix (*61 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP Set 123 call 235. Check that 102 is ringing. Take the call and check audio and display. Disable CFU on 235 using *60 prefix.	☐	☒	☐	
7A	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to another SIP user** **By local feature if applicable:** <u>Local</u>: On SIP set 235 enable CFU to SIP set 122 With 236 call 235. Check that 122 is ringing. Take the call and check audio and display. Disable CFU on 235. <u>Network</u>: On SIP set 235 enable CFU to IP Touch 103. With SIP Set 122 call 235. Check that 103 is ringing. Take the call and check audio and display. Disable CFU on 235.	☐	☒	☐	
7B	**By system feature:** <u>Local</u>: On SIP Set 235 enable CFU to SIP Set 122 using *61122 prefix (*61 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP Set 236 call 235. Check that 122 is ringing. Take the call and check audio and display. Disable CFU on 235 using *60 prefix. Network: On SIP Set 235 enable CFU to IP Touch 103 using *61123 prefix (*61 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP Set 122 call 235. Check that 103 is ringing. Take the call and check audio and display. Disable CFU on 235 using *60 prefix	☐	☒	☐	
8A	**Local call to SIP terminal in "forward on busy" (CFB) state:** **By local feature if applicable** On SIP Set 235 enable CFB to IP Touch 226 With 235 call the voice mail at 500 to make it busy. With SIP Set 236 call 235 which is busy. Check that 226 is ringing Take the call and check audio and display. Disable CFU on 235.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
8B	**By system feature:** On SIP Set 235 enable CFB to IP Touch 226 using *62226 prefix (*62 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With 235 call the voice mail at 500 to make it busy. With SIP Set 236 call 235 which is busy. Check that 226 is ringing Take the call and check audio and display. Disable CFB on 235 using *60 prefix.	☐	☒	☐	
9A	**Local call to SIP terminal in “forward on no reply” (CFNR)** **By local feature if applicable** On SIP Set 235 enable CFNR to IP Touch 226 With SIP Set 236 call 235. Check that 235 is ringing and don’t take the call, wait for time out (about 30 seconds). After time out expiration the 226 is ringing, take the call and check audio and display.	☐	☒	☐	
9B	**By system feature:** On SIP Set 235 enable CFNR to IP Touch 226 With SIP Set 236 call 235. Check that 235 is ringing and don’t take the call, wait for time out (about 30 seconds). After time out expiration the 226 is ringing, take the call and check audio and display.	☒	☐	☐	
10	**Call to busy user, Call waiting.** (Camp-on), local feature if applicable: With SIP Set 236 call other SIP Set 235 (multiline set) to make it busy, take the call and don’t hang up. With IP Touch 226 call 235 (on 235 camp-on feature is enabled). Check the Call waiting or ring back tones and display	☐	☒	☐	
11	**External call to SIP terminal.** Check that external call back number is shown correctly: With SIP Set 235 dial 9 + target MCDU number. Check that external is ringing and the external call number is shown correctly Take the call and check audio, display and call release.	☐	☒	☐	
12	**Calling Line Identity Restriction (CLIR): Local call to SIP terminal.** On IP Touch 226 enable mask Identity and call SIP Set 235 in order to hide 226 identity. Check that 235 is ringing, take the call and check that 226 identity is hidden.	☐	☒	☐	
13	Display: Call to free SIP terminal from IP Touch user with a name containing non-ASCII characters (eg éêêèè). Check caller display. Check that SIP set is ringing and check on its display that the characters are correctly printed.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
14	Display: Call from IP Touch to SIP which has the name containing non-ASCII characters, eg &@(#?+)=. Check caller display. Check that SIP set is ringing and check that the characters are correctly printed.	☐	☒	☐	
15	SIP set is part of a sequential hunt group **(1)**. Call to hunt group. Check call/release. With IP Touch 226 call the sequential hunt group MCDU number Check that 235 is ringing Take the call and don't hang up. And with IP Touch 226 call the sequential hunt group MCDU number Check that 235 is ringing Take the call and don't hang up. And with SIP Set 226 call the sequential hunt group MCDU number Check that 236 is ringing Take the call and don't hang up.	☐	☒	☐	
16	SIP set is part of a cyclic hunt group **(2)**. Call to hunt group. Check call/release. With IP Touch 226 call the cyclic hunt group MCDU number Check that 235 is ringing Take the call and hang up. And with 226 call the cyclic hunt group MCDU number Check that 235 is ringing Take the call and hang up. And with SIP Set 236 call the cyclic hunt group MCDU number Check that 235 is ringing Take the call and don't hang up.	☐	☒	☐	
17	SIP set is declared as a MultiSet. Call to main set and see if twin set rings. Take call with twin set. With IP Touch 226 call IP Touch 227 which is in MultiSet with SIP Set 235. Check that 227 and 235 both ringing. Take the call from 235 and check that 227 stop ringing. 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

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 several tests of this section. For more information refer to the appendix C.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1A	**Hold and resume with local feature** (if applicable) With 235 call 226 take the call, check audio and display. With 235 put 226 on hold check tones and display on both and resume the call. With 235 put 226 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 235 (multi-lines) call 226 and take the call. 226 will be put on hold when making second call to 227 Put 227 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 235 switch between 226 and 227 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 226 and only 235 and 227 are in call Check that 235 & 227 are still in a call, check display.	☐	☒	☐	
2	**Three party conferences initiated from OXO set** With 226 call 235, take the call and don't release it. With 226 call 227, take the call and don't release it too. With 226 start a conference? Check that 226, 227 and 235 are in conference. Check audio and display.	☐	☒	☐	
3	**Three party conferences initiated from SIP set with local feature** (if applicable) With 235 call 226 take the call and don't release it. With 235 call 227, take the call and don't release it too. With 235 start a conference by the local feature Check that 235, 226 and 227 are in conference. Check audio and display.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
3B	**Three party conferences initiated from SIP set with system feature** With 235 call 226 take the call and don't release it. With 235 call 227, take the call and don't release it too. With 235 start a conference by the OXO conference prefix Check that 235, 226 and 227 are in conference.	☒	☐	☐	
4	**Meet Me conference** With 235 call the Meet me Conference bridge dialing prefix 68 and follow instruction to open the bride. With 236 join the conference bridge by dialing prefix 69 and enter access code. With 226 join the conference bridge by dialing prefix 69 and enter access code. Check that 235, 236 and 226 are in conference.	☐	☒	☐	

8. 6 Call Transfer

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 *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 *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 transfer are not supported for SIP phones on OmniPCX Office.

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

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

8. 7 Attendant

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

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**SIP set call to attendant** From SIP set 235 dial “9” (attendant call prefix)Check audio and display	☐	☒	☐	
2	**2nd incoming call while in conversation with attendant** While SIP set 235 is in conversation with the attendant, from IP Touch 226 call 235 Answer the call and check audio and display	☐	☒	☐	
3	**SIP set call to attendant, attendant transfers to OXO set, semi-attended** From SIP set 235 dial “9” (attendant call prefix) and answer. Attendant transfer semi-attended to IP Touch 226 Answer the call and check audio and display	☐	☒	☐	
4	**SIP set call to attendant, attendant transfers to OXO set, attended** From SIP set 235 dial “9” (attendant call prefix) and answer Attendant transfer attended to IP Touch 226 Check audio and display	☐	☒	☐	
5	**OXO set calls to attendant, attendant transfers to SIP set, attended** From IP Touch 226 dial “9” (attendant call prefix) and answer Attendant transfer attended to SIP set 235 Check audio and display	☐	☒	☐	
6	**External ISDN Call to attendant, attendant transfers to SIP set, attended** ISDN incoming call to the attendant. From the attendant call SIP set 235 and transfer attended Check audio and display	☐	☒	☐	
7	**SIP set call to attendant, attendant transfers to External** From SIP set 235, dial “9” (attendant call prefix) and answer From the attendant, call an external ISDN destination and transfer semi-attended Answer and check audio and display.	☐	☒	☐	

8. 8 Voice Mail

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 235, 235 and OXO 226.

For these tests, DTMF sending (RFC 2833) has to be validated in order to use Voice Mail menu.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Password modification With SIP set 235 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 SIP set 235 call the Voice Mail at 500. Follow the instructions in order to send a voice message in SIP set 235 boxes. Check that the MWI on 235 is activated.	☐	☒	☐	
3	Message consultation With SIP set 236 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 236 after message cancellation.	☐	☒	☐	
4	SIP call to a OXO user forwarded to Voice Mail Forward the IP Touch 226 to Voice Mail by dialing *61500 (*61 prefix + <Voice Mail number>). With SIP set 235 call 226 and check that you are immediately forwarded to Voice Mail. Check that you can leave a message On 226 disable Voice Mail forwarding with *60 prefix.	☐	☒	☐	
5	OXO set call to a SIP user forwarded to Voice Mail Forward the SIP set 235 to Voice Mail by dialing *61500 (*61 prefix + <Voice Mail number>). With IP Touch 226 call 235 and check that you are immediately forwarded to Voice Mail. Check that you can leave a message On 235 disable Voice Mail forwarding with *60 prefix.	☐	☒	☐	

Notes:

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

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

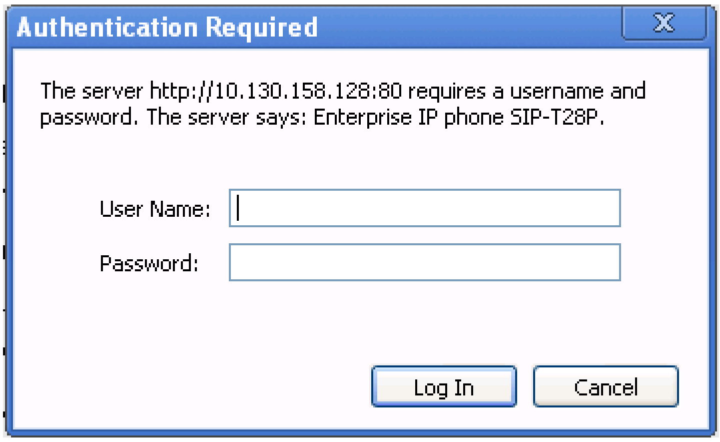
Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**OXO Reboot** Establish an incoming ISDN call with SIP set-1. Reboot the OXO. When the OXO is up again, re-establish an incoming ISDN call with SIPset-1 and check the audio.	☐	☒	☐	
2	**Ethernet link failure** Establish an incoming ISDN call with SIP set-1. Disconnect the Ethernet link of SIP set-1. Check that the incoming call is presented to the attendant. Reconnect the Ethernet link of SIP set-1. Re-establish an incoming ISDN call with SIP set-1 and check the audio.	☐	☒	☐	

9 Appendix A: AAPP member's Application Description

Yealink T28P

We can access the GUI of the phone with the IP address of the phone.The Yealink SIP-T28P represents the next generation VoIP phone designed for business users who need rich telephony features, a friendly user-interface and superb voice quality. Equipped with the TI TITAN chipset, it offers high-definition voice quality through a TI voice engine, HD handset, HD speaker.

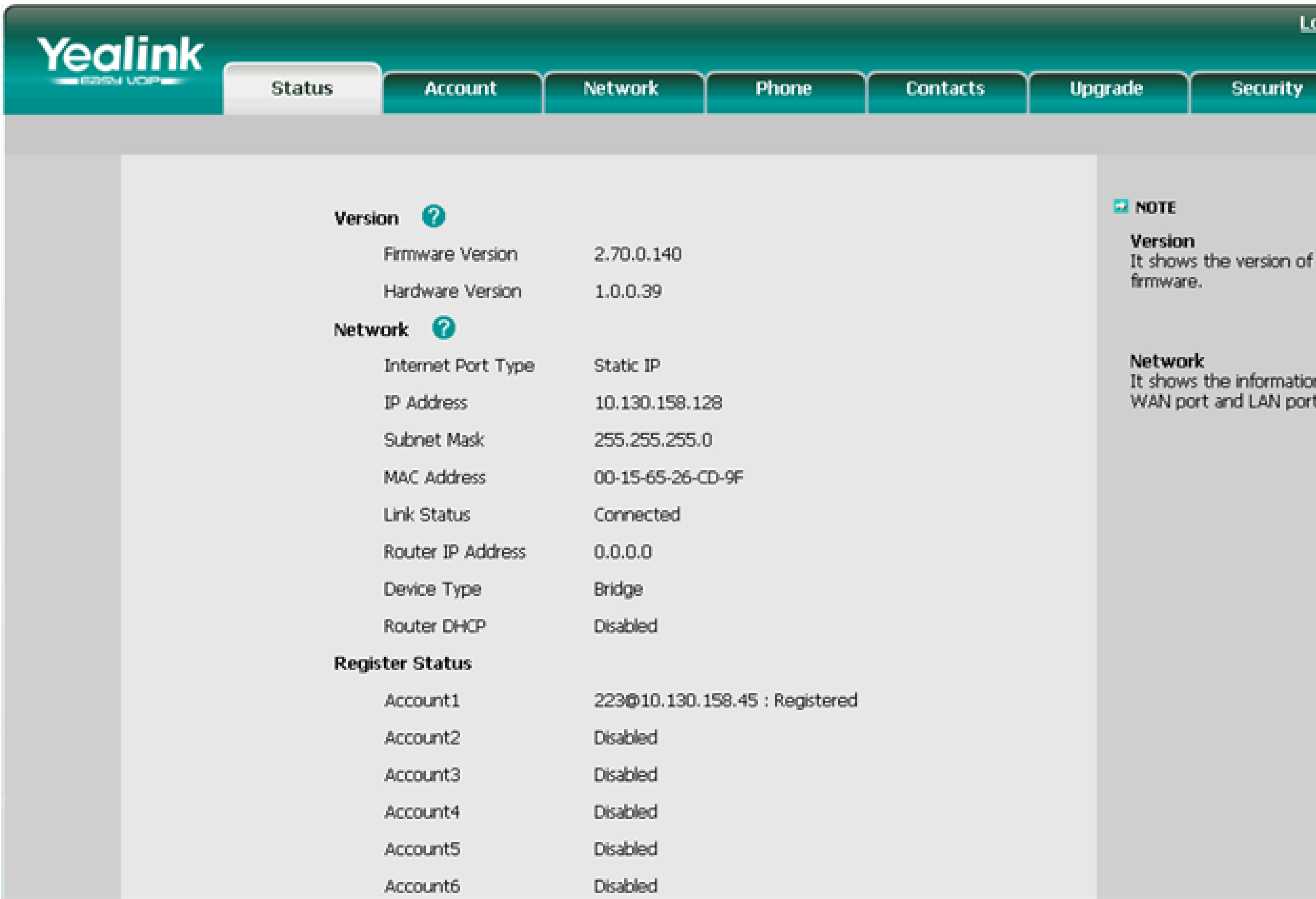
Login window



Username: admin
Password: admin

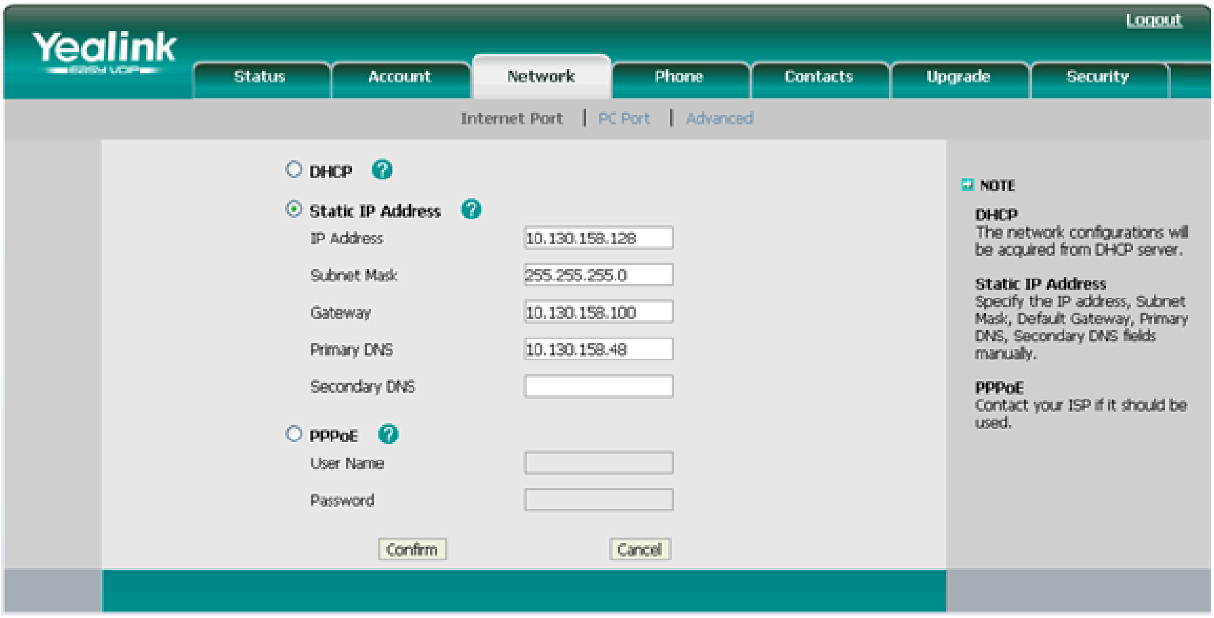
Phone status Screen

After successful authentication we can see the status of the phone.


Yealink
Status
Account
Network
Phone
Contacts
Upgrade
Security
Version
Firmware Version 2.70.0.140
Hardware Version 1.0.0.39
Network
Internet Port Type Static IP
IP Address 10.130.158.128
Subnet Mask 255.255.255.0
MAC Address 00-15-65-26-CD-9F
Link Status Connected
Router IP Address 0.0.0.0
Device Type Bridge
Router DHCP Disabled
Register Status
Account1 223@10.130.158.45 : Registered
Account2 Disabled
Account3 Disabled
Account4 Disabled
Account5 Disabled
Account6 Disabled
NOTE
Version
It shows the version of
firmware.
Network

10Appendix B: AAPP member's: Application Configuration Requirements

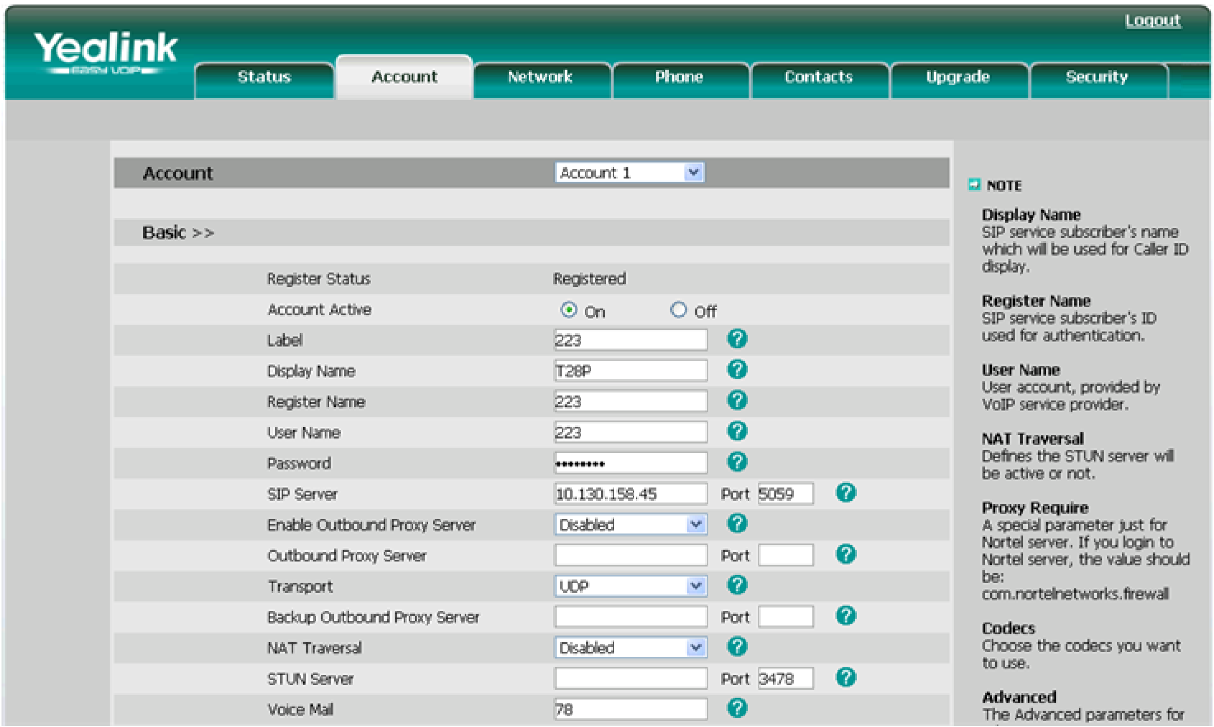
Network Settings section:



SIP account creation:

The SIP account should be configured and activated with the following parameters:

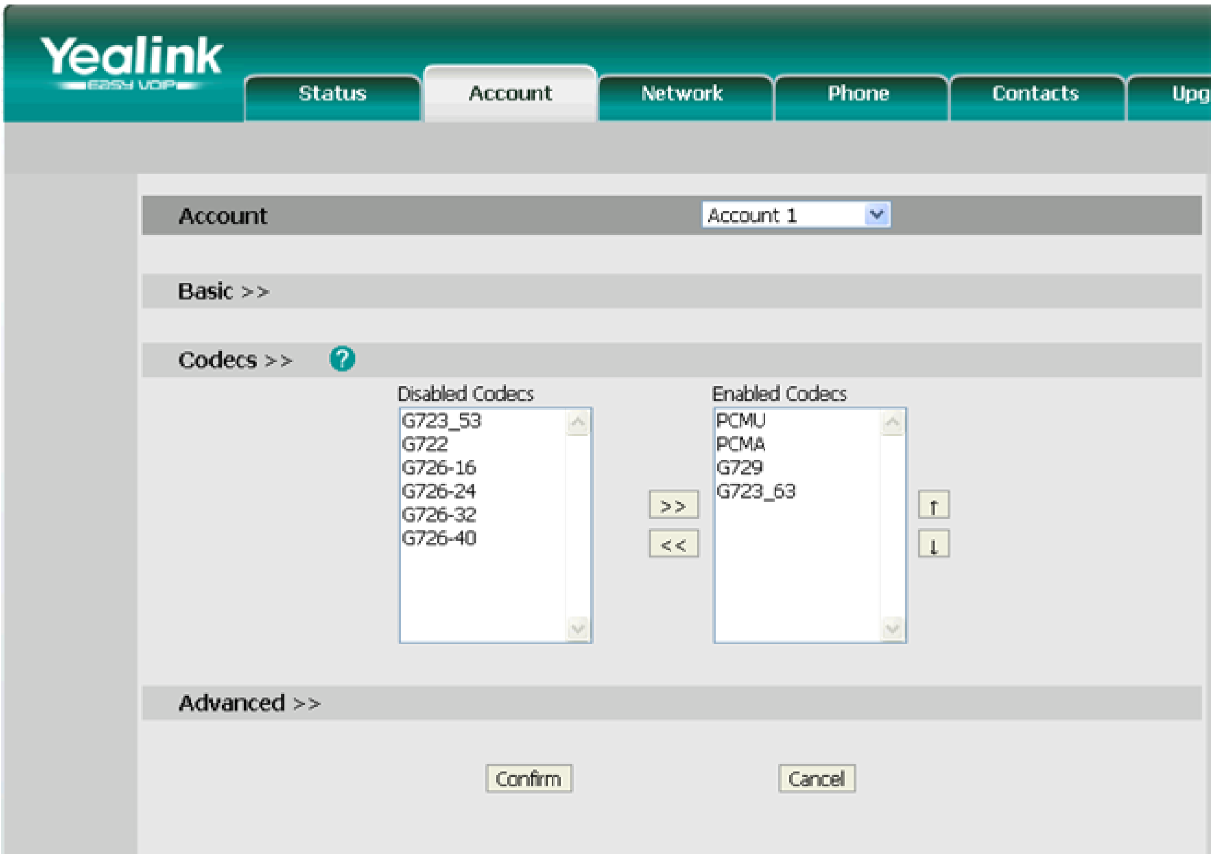
- Account name
- User name
- User password
- SIP domain (SIP proxy/registrar)
- System voice mail prefix



System voice mail Prefix :
Voice mail prefix is the maling prefix that should be specified to get access to OXO voice mail

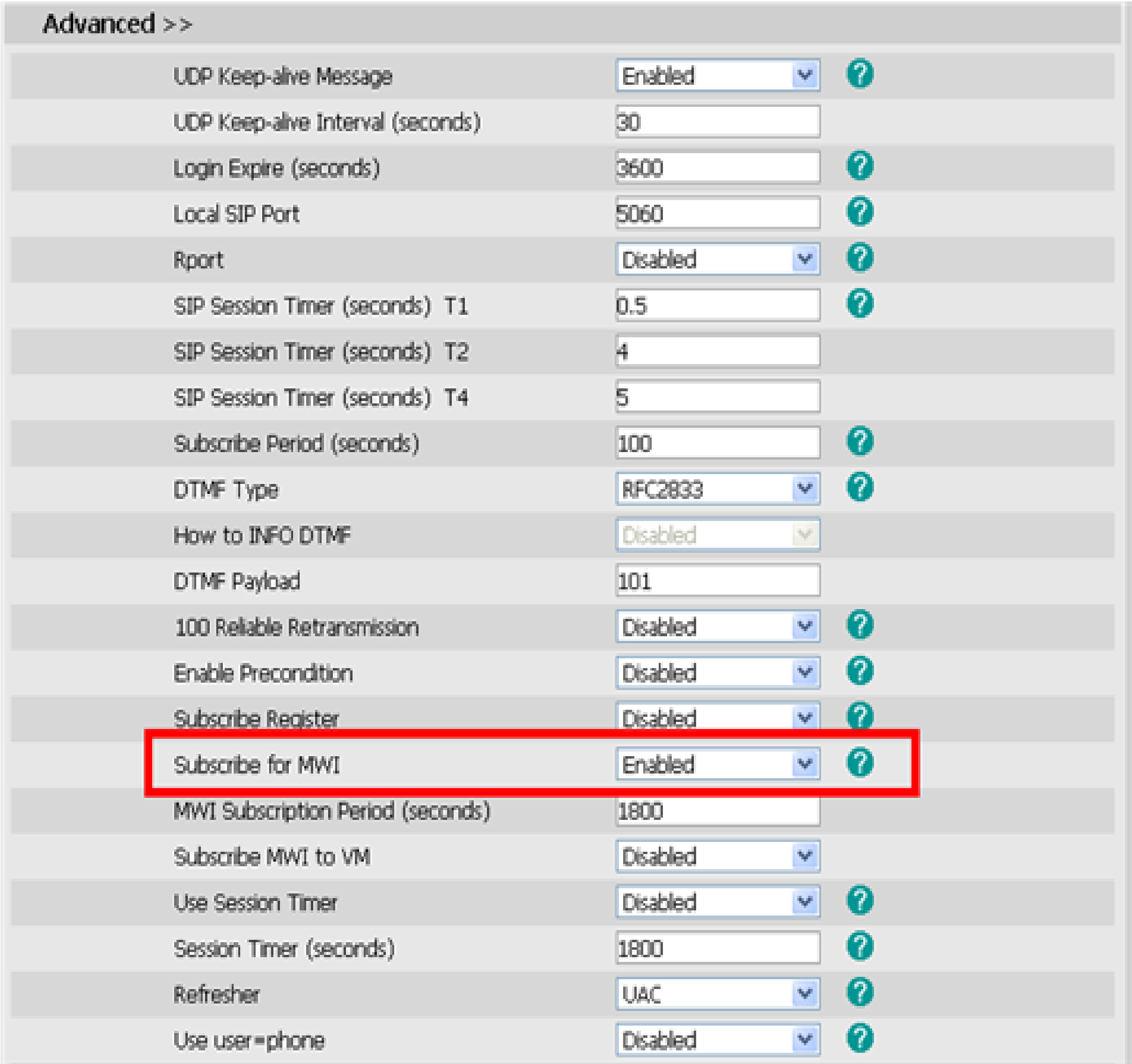
<u>Advanced parameters – codec options:</u>

The codec list should be configured in the advanced parameter section.



Voicemail MWI subcription:

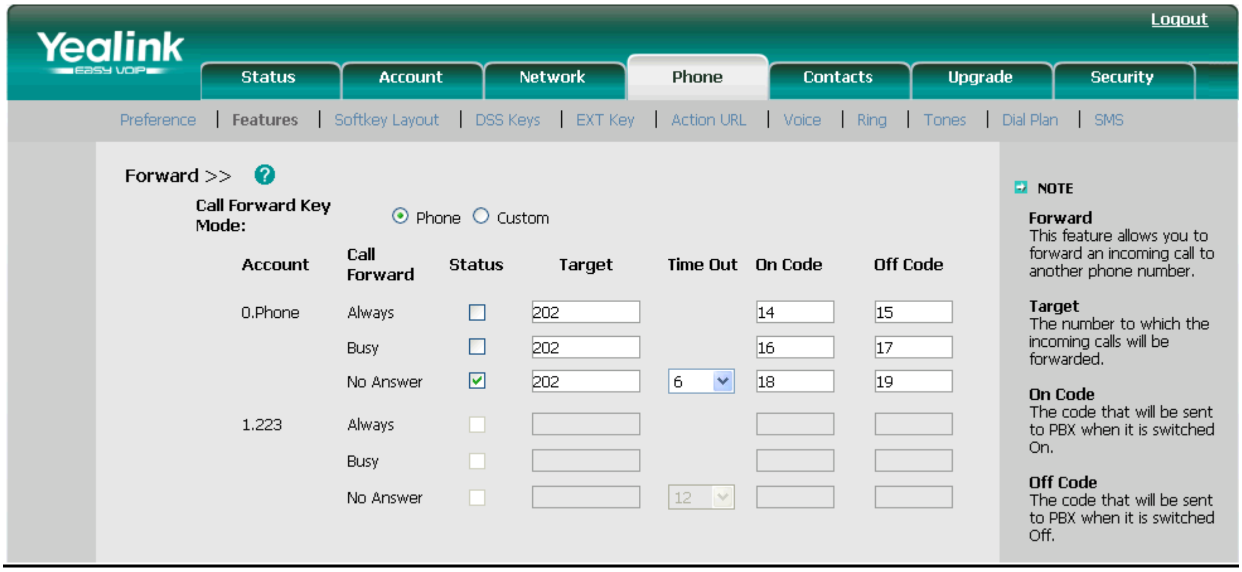
For message waiting indication we need to enable MWI subcription in advanced parmeters section.



Advanced >>

UDP Keep-alive Message	Enabled
UDP Keep-alive Interval (seconds)	30
Login Expire (seconds)	3600
Local SIP Port	5060
Rport	Disabled
SIP Session Timer (seconds) T1	0.5
SIP Session Timer (seconds) T2	4
SIP Session Timer (seconds) T4	5
Subscribe Period (seconds)	100
DTMF Type	RFC2833
How to INFO DTMF	Disabled
DTMF Payload	101
100 Reliable Retransmission	Disabled
Enable Precondition	Disabled
Subscribe Register	Disabled
Subscribe for MWI	Enabled
MWI Subscription Period (seconds)	1800
Subscribe MWI to VM	Disabled
Use Session Timer	Disabled
Session Timer (seconds)	1800
Refresher	UAC
Use user=phone	Disabled

Call forwarding

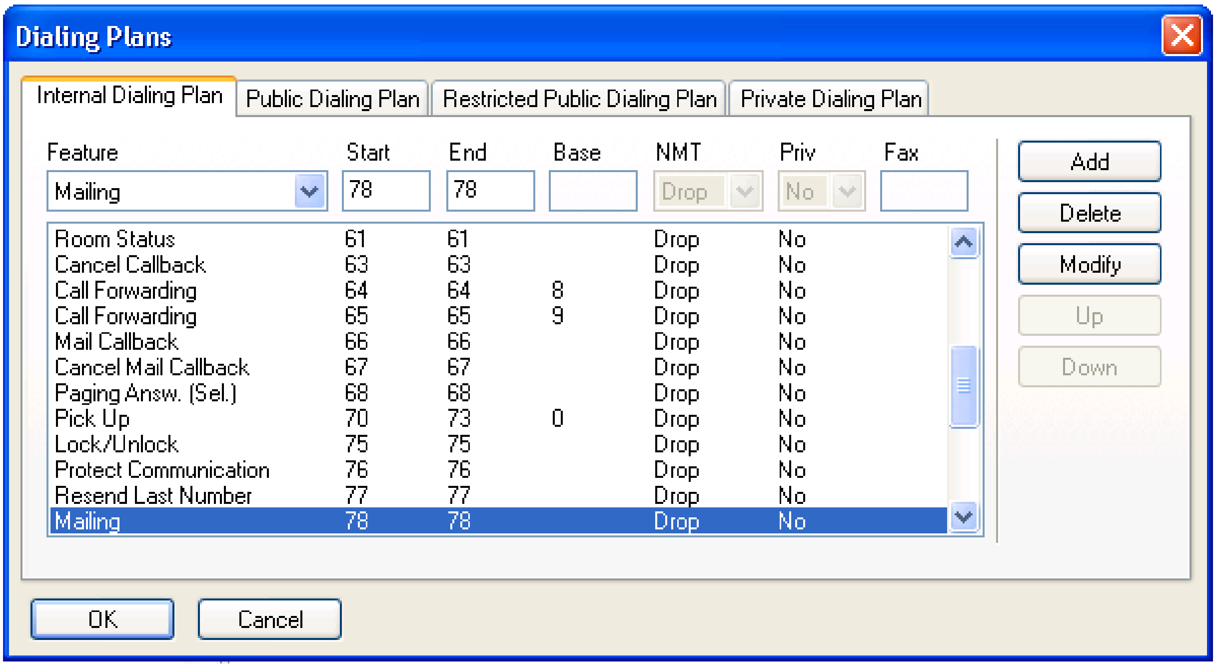


Do not Disturb

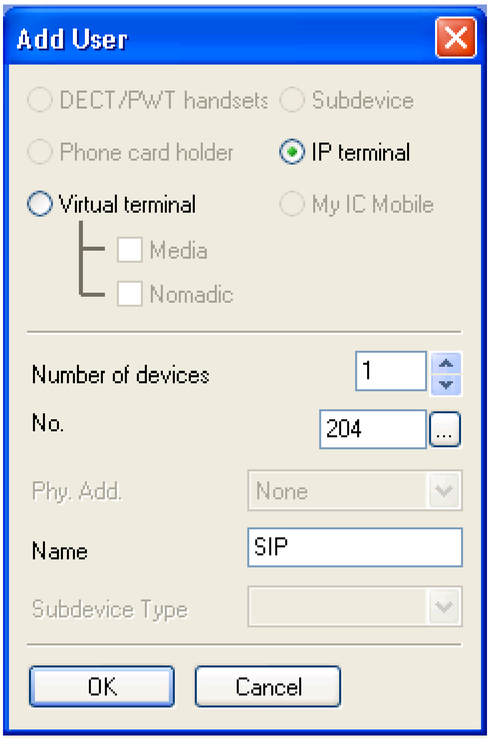
11 Appendix C: Alcatel-Lucent Platform: Configuration Requirements

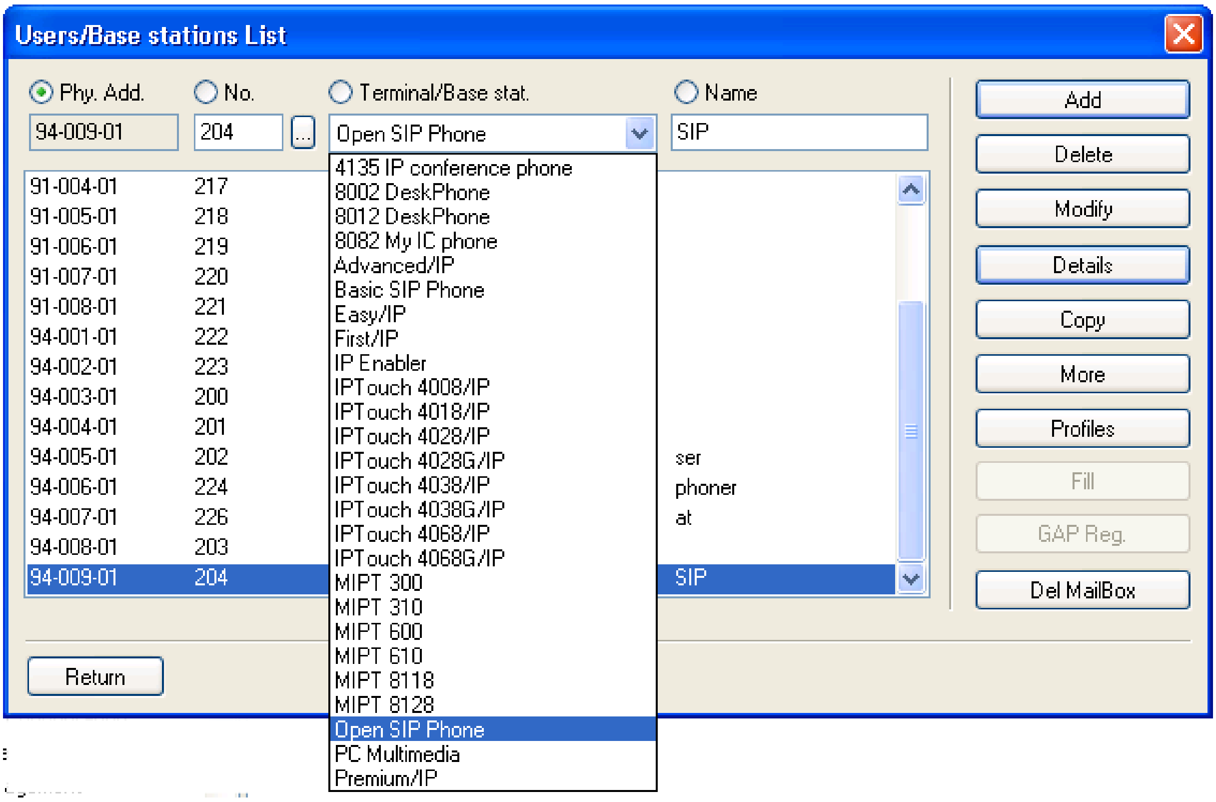
OXO Configuration

1. **Dialing Plan:**

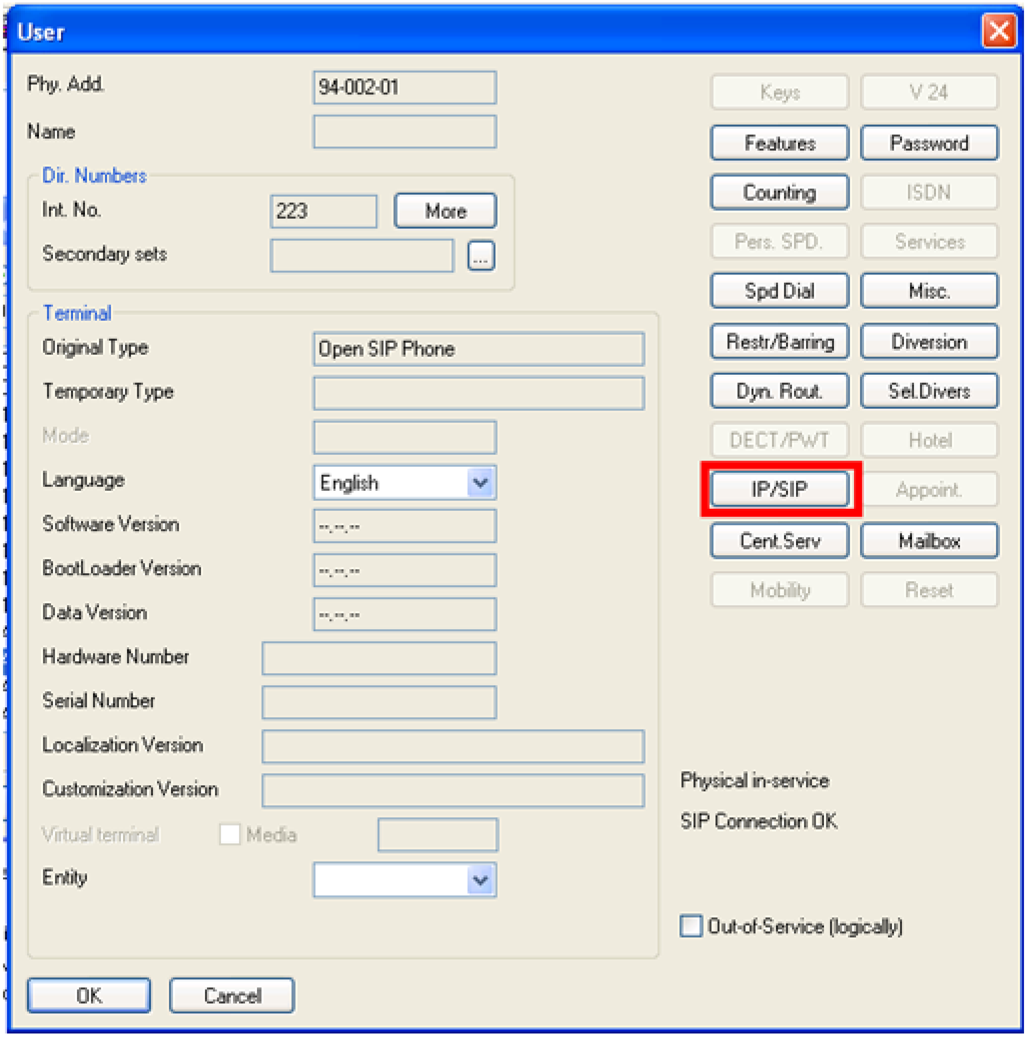


2. **User creation**

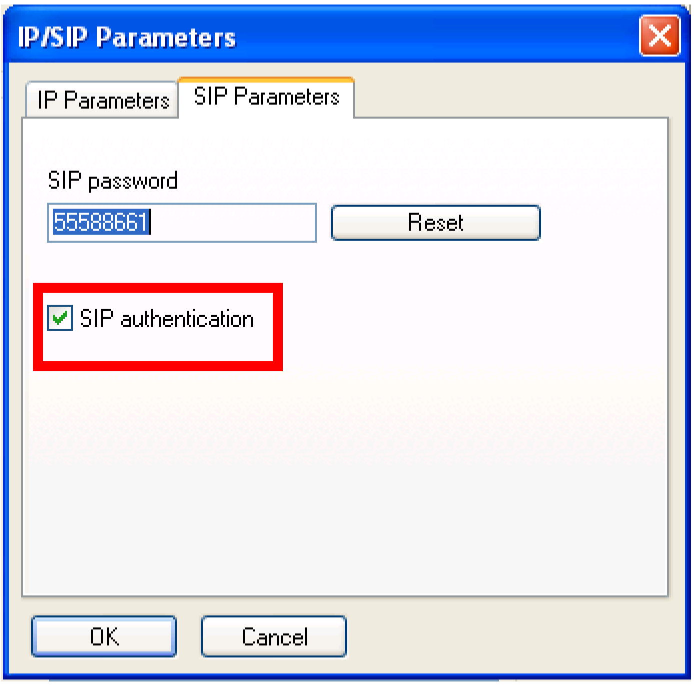




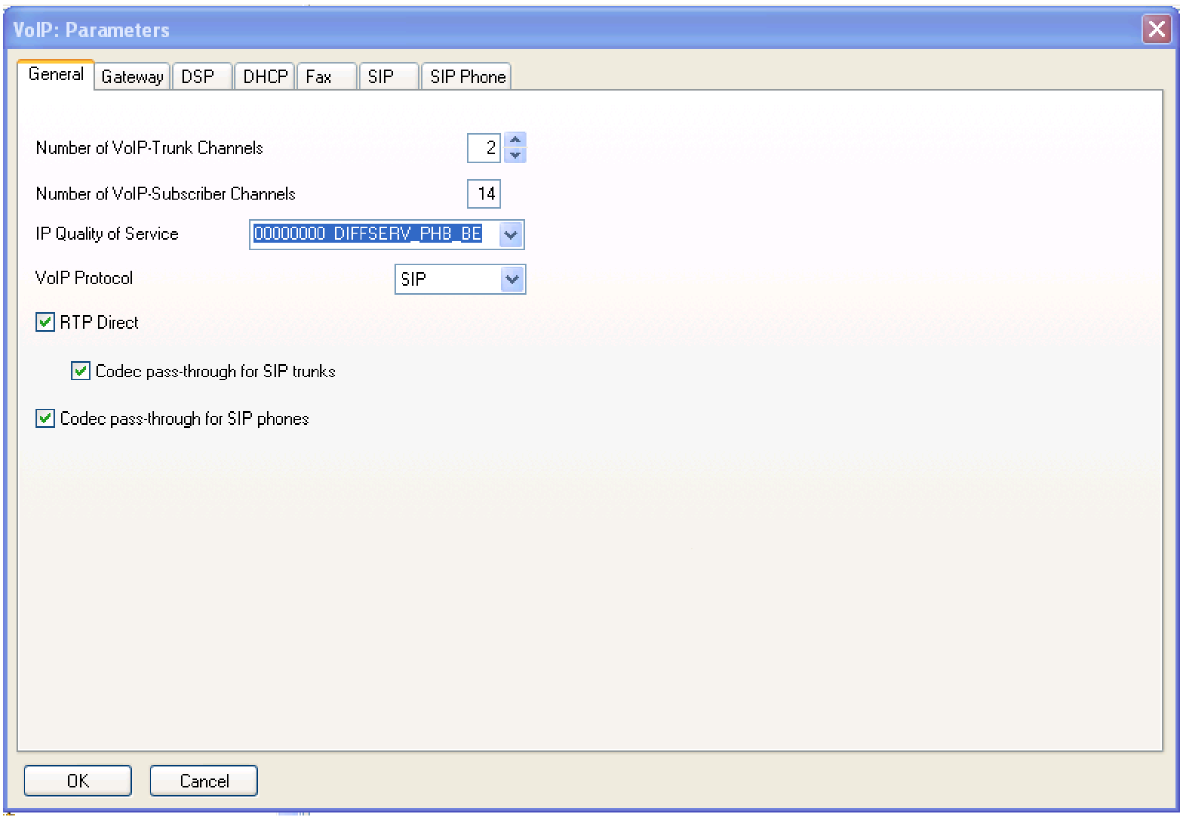
After modifying the IP terminal to Open SIP phone, click on details of the newly created open SIP extension

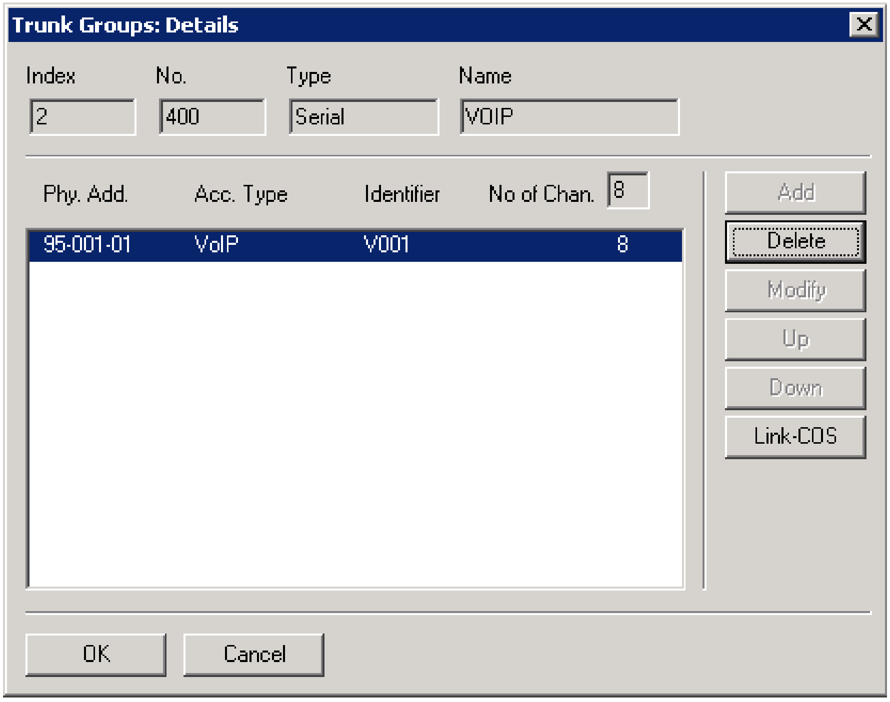


Click on IP/SIP to enable authentication

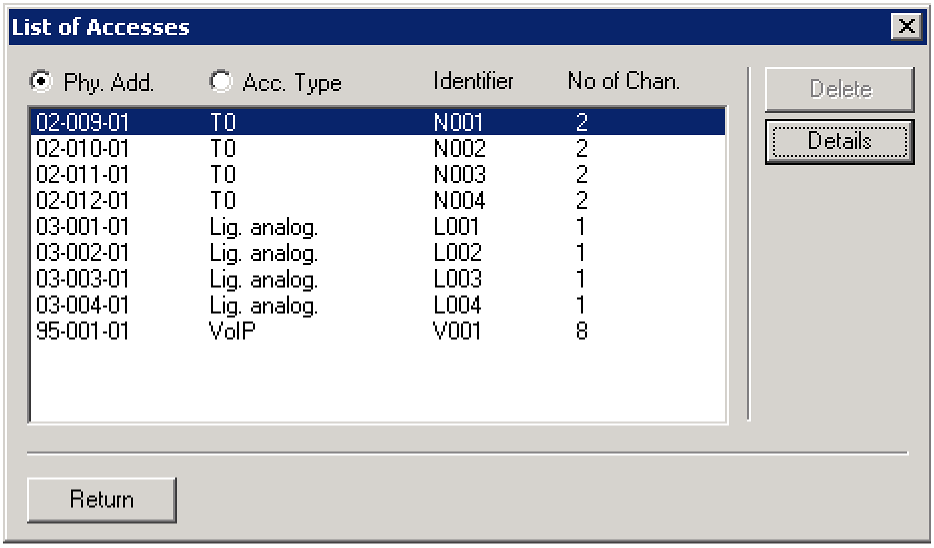


Trunk Configuration:

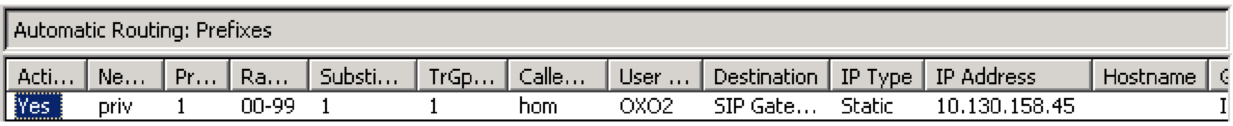




Trunk Access:



3. Network Call Configuration:



12Appendix D: AAPP member's escalation process

E-mail: support@yealink.com
Phone: +86 592 570 2000
Web address: www.yealink.com

13Appendix 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. 2Alcatel-Lucent.com

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

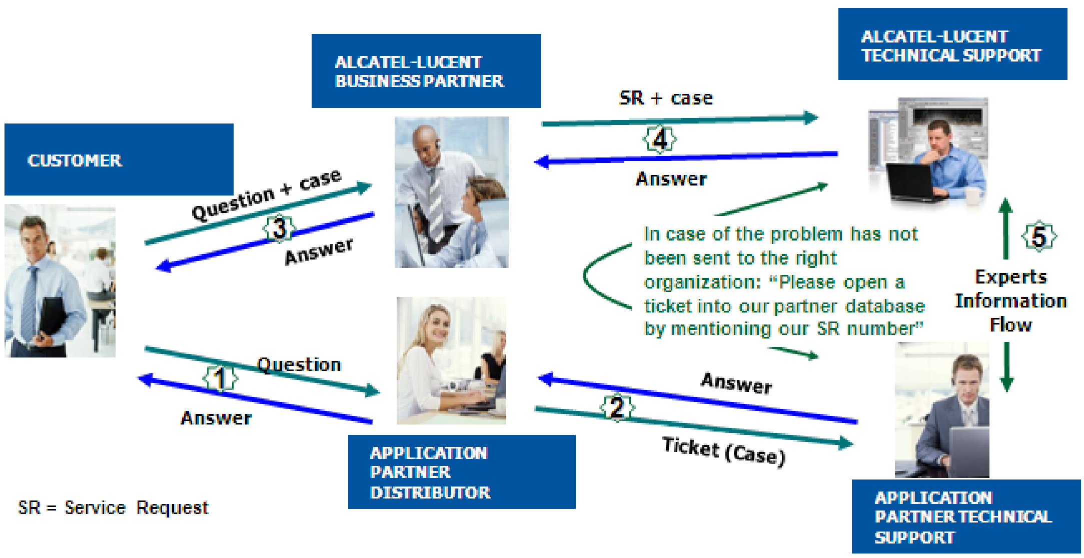
14Appendix F: AAPP Escalation process

14. 1Introduction

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

END OF DOCUMENT