Alcatel-Lucent Application Partner Program Inter-Working Report

Partner: YEALINK IP Phones
Application Type: Yealink SIP Phones
Application Name: T28P, T26P, T22P, T20P
Alcatel-Lucent Platform: OmniPCX Enterprise™

The product and version listed have been tested with the Alcatel-Lucent Communication Server and the version specified hereinafter. The tests concern only the inter-working between the Application Partner product and the Alcatel-Lucent Communication platforms. The inter-working report is valid until the Application Partner issues a new version of such product (incorporating new features or functionality), or until Alcatel-Lucent issues a new version of such Alcatel-Lucent product (incorporating new features or functionality), whichever first occurs.

Tests identification

Date of the tests	April 2012

Alcatel-Lucent's representative	Chen Nixon
Partner's representative	Pablo Wang

Alcatel-Lucent Communication Platform	OmniPCX Enterprise
Alcatel-Lucent compatibility release	R10.0 **J1.410.40.c**
Partner's application version	**2.60.0.120/6.60.0.120/7.60.0.120/9.60.0.120**

Author(s): Chen Nixon, Florian Residori
Reviewer(s): Denis Lienhart

Historic

Edition 1: creation of the document – *April 2012*
Edition 2: update to take into account T28P and T26P – *June 2012*

Test results

☐ Passed ☐ Refused ☐ Postponed

☑ Passed with restrictions

Refer to the section 4 for a summary of the test results.

Company Contact Information

Contact name:	Pablo Wang
Title:	Business development manager
Address 1:	4th-5th Floor, South Building, No.63 Wanghai road, 2nd Software Park
Address 2:	
City:	Xiamen
State:	
Zip:	361006
Country:	China
Country code:	
Phone:	+86 592 570 2000
Fax:	+86 592 570 2455
Web address:	www.yealink.com
E-mail:	pablo@yealink.com

TABLE OF CONTENTS

1 INTRODUCTION 5

2 APPLICATION INFORMATION 6

3 TEST ENVIRONMENT 8

3.1 HARDWARE CONFIGURATION 9
3.2 SOFTWARE CONFIGURATION 10

4 SUMMARY OF TEST RESULTS 11

4.1 SUMMARY OF MAIN FUNCTIONS SUPPORTED 11
4.2 SUMMARY OF PROBLEMS 12
4.3 SUMMARY OF LIMITATIONS 12
4.4 NOTES, REMARKS 12

5 TEST RESULT TEMPLATE 13

6 TESTING 14

6.1 GENERIC TESTS 14
6.1.1 Phone initialization, SIP registration and authentication 14
6.1.2 Audio codecs negotiations 15
6.1.3 Defence 16
6.1.4 Basic call 17
6.1.5 Local telephonic features 19
6.1.6 Other features 21
6.2 HOTEL / HOSPITAL TESTS 22
6.2.1 Provisioning 22
6.2.2 Check-in and check-out 23
6.2.3 Do not disturb 24
6.2.4 Forward 25
6.2.5 Voice mail 26
6.2.6 Wake up 27
6.2.7 Change room state 28
6.2.8 Mini bar 28
6.2.9 Calls 29
6.2.10 Multi line 30
6.2.11 Multi occupation 32
6.2.12 Additional Tests. 33

7 APPENDIX A: PARTNER APPLICATION DESCRIPTION 34

8 APPENDIX B: PARTNER APPLICATION: CONFIGURATION REQUIREMENTS 37

9 APPENDIX C: ALCATEL-LUCENT PLATFORM: CONFIGURATION REQUIREMENTS 39

10 APPENDIX D: PARTNER ESCALATION PROCESS 46

11 APPENDIX E: AAPP PROGRAM 47

11.1 ALCATEL-LUCENT APPLICATION PARTNER PROGRAM (AAPP) 47
11.2 ALCATEL-LUCENT.COM 47

12 APPENDIX F: AAPP ESCALATION PROCESS 48

12.1 INTRODUCTION 48
12.2 ESCALATION IN CASE OF CERTIFIED APPLICATION/PRODUCTS 49
12.3 ESCALATION IN CASE OF NON-CERTIFIED APPLICATION/PRODUCT 50
12.4 TECHNICAL SUPPORT ACCESS 51

1 Introduction

The goal of these tests is to qualify an external application as an Alcatel-Lucent Application Partner Program solution for the Alcatel-Lucent Communication Platform.

The scope of the tests is the interoperability of the application with the Alcatel-Lucent Communication Platform. It covers a basic or complex inter-working to ensure that services requested by the application and provided by the Communication Platform (and/or conversely) are properly completed.

These tests do not check the functional achievement of the application as well as they do not cover load capacity checks, race conditions and generally speaking any real customer's site conditions.

2 Application information

Application type:	VOIP SIP Phone
Application commercial name:	T28P, T26P, T22P, T20P
Application version:	**2.60.0.120** for T28P, **6.60.0.120** for T26P, **7.60.0.120** for T22P, **9.60.0.120** for T20P
Interface type:	SIP/Ethernet

Interface version (if relevant):

Brief application description:

Yealink, the most popular Chinese brand of IP Phone in Western market, has been well known by customers and enjoys a worldwide fame. As the navigator of IP communication industry, Yealink focuses on design and manufacture of IP phones with low carbon and high quality to provide excellent and efficient technical supports, helping clients from all over the world to their business success.

YEALINK T28P SIP Phone:

It has Message, Headset, Transfer, Redial, Speaker, Mute and Volume + and –, hold and Conference key and 10 Programmable keys.

YEALINK T26P SIP Phone:

It has Message, Headset, Transfer, Redial, Speaker, Mute and Volume + and –, hold and Conference key and 10 Programmable keys.

YEALINK T22P SIP Phone:

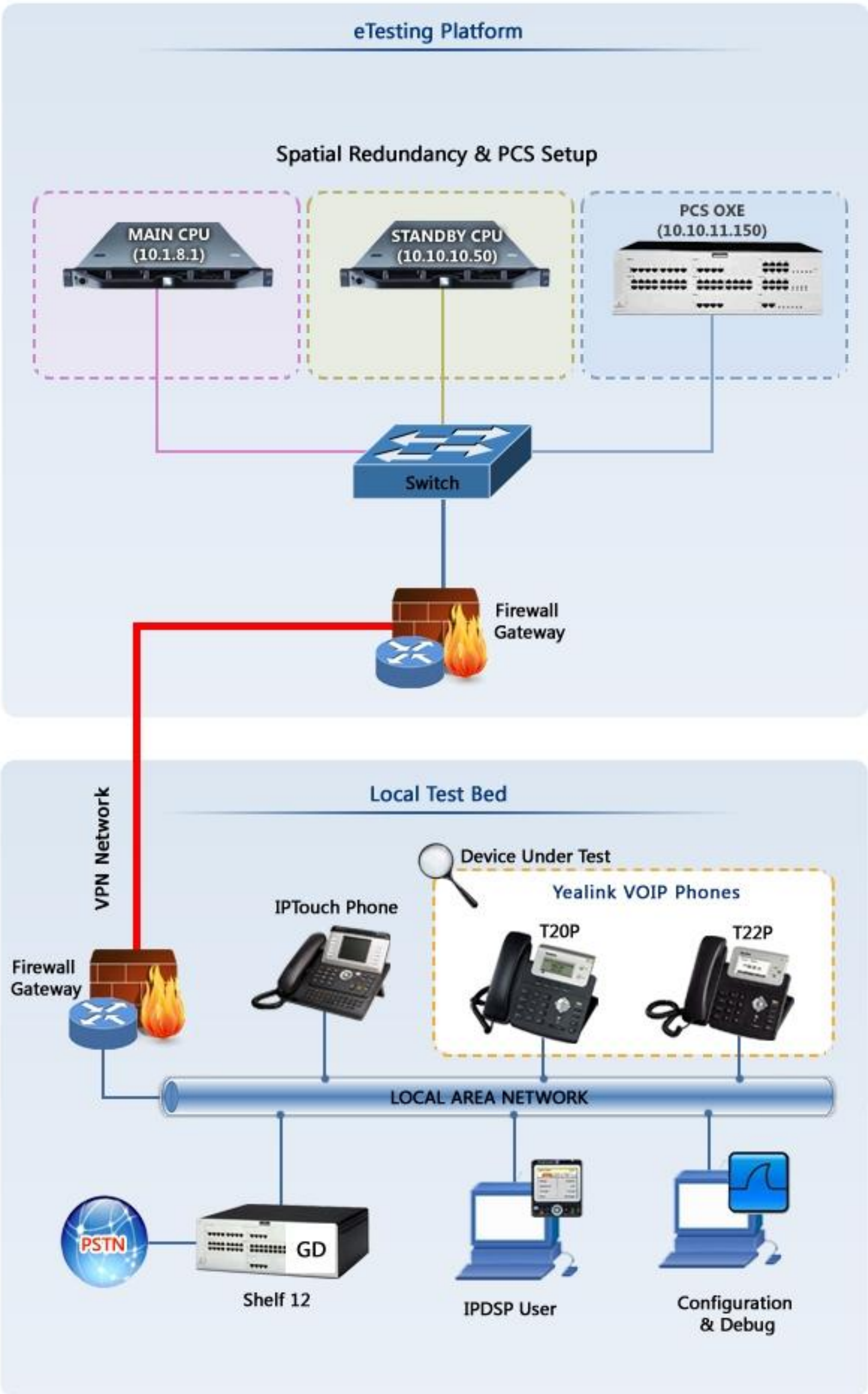
It has Message, Headset, Forward, Redial, Speaker, Mute and Volume + and – key.

YEALINK T20P SIP Phone:

It has Message, Headset, Transfer, Redial, Speaker, Mute and Volume + and –, hold and Conference key.

3 Test environment

Figure 1 **Test Environment**



3.1 Hardware configuration

- **Alcatel-Lucent Communication Platform**:

Node1:

- OmniPCX Enterprise common hardware
- Duplicated call servers : 10.1.8.1 and 10.10.10.50
- Passive Call Server (PCS) : 10.10.11.150
- Spatial redundancy (Different IP sub networks)
- Release : R10.0 - J1.410.40.c
- One media gateway:

Setup Details:

Setup Information	
Module	**Details**
Primary OXE	10.1.8.1
Secondary OXE	10.10.10.50
PCS OXE	10.10.11.150
Voicemail No	31300
Attendant prefix	***
IPTouch number	1281,1282
YEALINK SIP Phone number	1291 & 1292
Thomson TB30	1296
Analog & UA	1290 & 1298

Crystal 0

```
+-------------------------------------------------------------------+
| Cr  | cpl| cpl type    | hw type    |  cpl  state   |  coupler ID      |
|----|----|-------------|------------|---------------|------------------|
|  0 |  6 |       CPU_CS|----------- |   IN SERVICE |   BAD PCMS CODE |
|  0 | 10 |       CPU_CS|----------- |   IN SERVICE |   BAD PCMS CODE |
+-------------------------------------------------------------------+
```

Crystal 12

```
+-------------------------------------------------------------------+
| Cr  | cpl| cpl type    | hw type    |  cpl  state   |  coupler ID      |
|----|----|-------------|------------|---------------|------------------|
| 12 |  0 |           GD|----------- |   IN SERVICE |    NO PCMS CODE |
| 12 |  1 |        UAI 4|----------- | REG NOT INIT |    NO PCMS CODE |
| 12 |  2 |       PRA T1|----------- | REG NOT INIT |    NO PCMS CODE |
| 12 |  3 |       PRA T2|----------- |   IN SERVICE |    NO PCMS CODE |
| 12 |  4 |       MIX484|----------- |   IN SERVICE |    NO PCMS CODE |
| 12 |  5 |       PRA T2|----------- |   IN SERVICE |    NO PCMS CODE |
| 12 |  6 |        BRA 4|----------- | REG NOT INIT |    NO PCMS CODE |
| 12 |  7 |       PRA T1|----------- | REG NOT INIT |    NO PCMS CODE |
+-------------------------------------------------------------------+
```

Node2:

- OmniPCX Enterprise common hardware
- Single CPU : 10.10.10.50
- Release : R10.0 - J1.410.40.c
- One media gateway:

Crystal 0

```
+-----------------------------------------------------------------------+
| Cr  | cpl| cpl type     | hw type     |  cpl  state   |  coupler ID     |
|----|----|------------|-----------|--------------|-----------------|
|  0 |  6 |       CPU_CS|---------- |   IN SERVICE |   BAD PCMS CODE |
|  0 | 10 |       CPU_CS|---------- |   IN SERVICE |   BAD PCMS CODE |
+-----------------------------------------------------------------------+
```

Crystal 12

```
+-----------------------------------------------------------------------+
| Cr  | cpl| cpl type     | hw type     |  cpl  state   |  coupler ID     |
|----|----|------------|-----------|--------------|-----------------|
| 12 |  0 |           GD|---------- |   IN SERVICE |    NO PCMS CODE |
| 12 |  1 |        UAI 4|---------- | REG NOT INIT |    NO PCMS CODE |
| 12 |  2 |       PRA T1|---------- | REG NOT INIT |    NO PCMS CODE |
| 12 |  3 |       PRA T2|---------- |   IN SERVICE |    NO PCMS CODE |
| 12 |  4 |       MIX484|---------- |   IN SERVICE |    NO PCMS CODE |
| 12 |  5 |       PRA T2|---------- |   IN SERVICE |    NO PCMS CODE |
| 12 |  6 |        BRA 4|---------- | REG NOT INIT |    NO PCMS CODE |
| 12 |  7 |       PRA T1|---------- | REG NOT INIT |    NO PCMS CODE |
+-----------------------------------------------------------------------+
```

- **OmniPCX Entreprise**:
  - CS (Call Server Processing Unit)
  - GD (Gateway driver processing Unit
  - PRA T2 (ISDN Access)
  - MIX 4/8/4 (ISDN T0, digital & analog interfaces)
  - UA digital and analog sets

- **AHL interface**:
  - TCP/IP

3.2 Software configuration

- **Alcatel Communication Platform**: OmniPCX Enterprise **R10.0 - J1.410.40.c**

- **Partner Application** : YEALINK SIP Phone **V2.60.0.120** for T28P, **V6.60.0.120** for T26P, **V7.60.0.120** for T22P, **V9.60.0.120** for T20P model

4 Summary of test results

4.1 Summary of main functions supported

Features	Status	Comments
Initialization including network configuration	OK	.
Defences	OK	
SIP registration	OK	
SIP authentication	OK	
Voice over IP and RTP codec support	OK	
Basic calls	OK	
Local Telephonic Features	OK	
Date and time display and update	OK	
Hotel features		
Registration as administrative or room set	OK	
Check-in and out	OK	
OXE hotel telephonic features		
Do not disturb	OK	
Forward	OK	“On busy” and “On busy / no answer” not supported
Wake-up	OK	
Voice Mail	OK_But	“Old message” display is not correct
Call back	OK	
Conference	OK_But	OXE Suffix not supported. Phone local feature to be used.
Enquiry call	OK_But	OXE Suffix not supported. Phone local feature to be used.
Broker call	OK_But	OXE Suffix not supported. Phone local feature to be used.
Room state management	OK	
Single room / suite management	OK	
Basic calls	OK	
Multi line	OK	
Mini bar	OK	
Multi occupation	OK	

4.2 Summary of problems

- There is no disconnecting tone available in the Yealink phones, when we disconnect the call at other side. SR: 1-134567433

4.3 Summary of limitations

- When SIP user disconnects the IPTouch User after an OXE CPU switchover, the call is not disconnectied. SR 1-134653875
- In the display of the Yealink phone, the number of New & Old messages are one at the same (always the number of new messages), irrespective of the actual number of old messages.
- In DND mode, the Yealink phone is not ringing when we call from the other user, but in the display of the phone this call is notified as missed call.
- Suffix 8 used to leave a message on the voicemail doesn't work on Hotel Guests. SR : 1-118746681

4.4 Notes, remarks

Environment setup for Generic Test:

- The phone is configured as SIP Extension
- It is configured as ROOM + Multi occupancy
- it is NOT checked in with a GUEST number in the HOTMENU

Environment setup for Hotel Tests

- The phone is configured as SIP Extension
- It is configured as ROOM + Multi occupancy
- And it is CHECKED- IN with a GUEST number in the HOTMENU

5 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 <u>and the expected result</u>
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, <u>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</u>
Comment: to be filled in with any relevant comment. Mandatory in case a test has failed especially the reference number of the issue.

6 Testing

6.1 Generic tests

These tests check the phones behavior for generic SIP phone features like initializations and registrations, audio parameters and voice quality, defenses, basic calls and telephonic features.

6.1.1 Phone initialization, SIP registration and authentication

These tests check that the phones are able to register to the OXE with and without SIP authentication, by using DNS or not, DHCP mode or static IP addressing, 802.1p/Q VLAN tagging and priority.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**SIP set registration to OXE in DHCP mode** The phone is configured to get its IP address by DHCP	☐	☒	☐	
2	**SIP set registration to OXE in static IP addressing mode** The phone is configured to use IP static parameters	☐	☒	☐	
3	**SIP set registration to OXE using a DNS or alternate proxy** The phone is configured to use a domain name as registrar / proxy server address. The DNS IP addresses are the OXE CPU address. In case of alternate proxy possibilities, the main and alternate proxy addresses are the OXE CPU address. Tests are performed when first Call Server is active and then when second Call Server is active	☐	☒	☐	The DNS method & Alternate proxy method has issues in disconnecting the call from the SIP user after the switchover. SR 1-134653875
4	**SIP set registration to OXE using SIP digest authentication** SIP digest authentication is activated on OXE and phone side. Check also that outgoing call is authenticated.	☐	☒	☐	

6.1.2 Audio codecs negotiations

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

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**The phone is configured to offer G711Alaw, G723 and G729 (in this priority order). The OXE is configured to use G711Alaw.** Check that for an incoming and outgoing call, the negotiated codec is G711 A law	☐	☒	☐	
2	**The phone is configured to offer G711Alaw, G723 and G729 (in this priority order). The OXE is configured to use G723.** Check that for an incoming and outgoing call, the negotiated codec is G723	☐	☒	☐	
3	**The phone is configured to offer G711Alaw, G723 and G729 (in this priority order). The OXE is configured to use G729.** Check that for an incoming and outgoing call, the negotiated codec is G729	☐	☒	☐	
4	**The phone is configured to offer G711Alaw. The OXE is configured to use G711Alaw.** Check that for an incoming and outgoing call, the negotiated codec is G711 A law	☐	☒	☐	
5	**The phone is configured to offer G723. The OXE is configured to use G723**. Check that for an incoming and outgoing call, the negotiated codec is G723.	☐	☒	☐	
6	**The phone is configured to offer G729. The OXE is configured to use G729.** Check that for an incoming and outgoing call, the negotiated codec is G729.	☐	☒	☐	
7	Repeat previous 6 tests by changing A law to µlaw	☐	☒	☐	
8	**Codec selection. The phone and OXE do not have any common codec.** For example, the phone is configured to offer G723 and the OXE to use only G729 (use of IP domains). Check that for an incoming and outgoing call is properly rejected.	☐	☒	☐	

6.1.3 Defence

These tests check the phones defenses against perturbations and OXE Call Servers switch over.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**OXE Call Server CPU switch over while SIP phone in idle.** Check the SIP phone behavior after a switch from the OXE main to standby CPU. The phone must be able to make and receive a call after the switch over.	☐	☒	☐	
2	**OXE Call Server CPU switch over while SIP phone in conversation with an IPTouch.** Check the SIP phone behavior after a switch from the OXE main to standby CPU. The call is still active. The phone can make and receive a second call and switch from one to another. After on hook, the phone must be able to make and receive a call after the switch over.	☐	☐	☒	When SIP user disconnects the IPTouch User, the call is not disconnecting after switchover. SR 1-134653875
3	**OXE Call Server reboot while SIP phone in idle.** Check the phone behavior when the OXE Call Server reboots (without standby CPU). As soon as the Call Server is running again, the phone is able to make and receive a call.	☐	☒	☐	
4	**OXE Call Server reboot while SIP phone in conversation with an IPTouch.** Check the phone behavior when the OXE Call Server reboots (without standby CPU). The call is released. As soon as the Call Server is running again, the phone is able to make and receive a call.	☐	☒	☐	

6.1.4 Basic call

These tests check the phone behavior during basic incoming and outgoing calls from and to different kind of phone set types (SIP, IPTouch, UA) with different call releases (during ringing, by caller, by callee) and with or without a second incoming call.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Call from and to a SIP phone.** The phone calls a SIP phone. The phone is called by a SIP phone. In both cases, check the display and audio during all steps (dialing, ring back tone, conversation, and release).	☐	☒	☐	There is no disconnecting tone available in the Yealink phones, when we disconnect the call at other side. SR: 1-134567433
2	**Call from and to an IPTouch.** Same as **1** but with an IPTouch.	☐	☒	☐	
3	**Call from and to an UA phone.** Same as **1** but with an UA phone.	☐	☒	☐	
4	**Incoming call released by the caller during ringing.** The caller releases the incoming call to the phone before the callee takes the call.	☐	☒	☐	
5	**Outgoing call released by the caller during ringing.** The caller releases the outgoing call from the phone before the callee takes the call.	☐	☒	☐	
6	**Incoming call rejected by the callee during ringing**. The callee rejects the incoming call to the phone during ringing.	☐	☒	☐	
7	**Outgoing call rejected by the callee during ringing.** The callee rejects the outgoing call from the phone during ringing.	☐	☒	☐	
8	**Call released by the phone.** The phone releases the call after a conversation period.	☐	☒	☐	
9	**Call released by the other phone.** The other phone releases the call after a conversation period.	☐	☒	☐	
10	Call from and to an external number (T0/T2) Call is properly established.	☐	☒	☐	
11	**Call from and to an attendant** Call is properly established.	☐	☒	☐	
12	**Incoming external call (T0/T2 for example) to an attendant phone set which transfers the call to the phone. Transfer is done while the phone is ringing but also after this one has picked up the call (using the attendant soft key or going on hook).** Call is properly established.	☐	☒	☐	
13	**Outgoing call from a phone to an attendant with transfers to an external call (T0/T2 for example).** Call is properly established.	☐	☒	☐	
14	**Dialing break**	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
	The phone starts dialing another phone number. Before the end, the dialing is stopped. Check that the phone comes back to idle state after the timeout expires.				

6.1.5 Local telephonic features

These tests check the phone behavior during phone local telephonic feature use like forward, on hold, transfer, voice mail interactions, conference.
These features are either activated through the phone web manager or with phone keys (if available).

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Immediate forward to another phone**. The phone is forwarded to another phone. Call The phone and check that the call is presented on the third phone and can be taken by this one.	☒	☐	☐	Local forward feature is not supported
2	**Forward on no answer to another phone.** The phone is forwarded on no answer to another phone. Call the phone and check that the call is presented on analog phone. Do not take the call and wait for the call to be presented on the third phone. Take the call on the third phone. Check also the call can be picked up before the call is forwarded.	☒	☐	☐	Local forward feature is not supported
3	**Forward on busy to another phone.** The phone is forwarded on busy to another phone. While the phone is already in conversation, call The phone and check that the call is presented. Do not take the call and wait for the call to be presented on the third phone. Take the call on the third phone.	☒	☐	☐	Local forward feature is not supported
4	**Forward on busy / no answer to another phone.** The phone is forwarded on busy / no answer to another phone. While the phone is already in conversation, call The phone and check that the call is presented. Do not take the call and wait for the call to be presented on the third phone. Take the call on the third phone. Call The phone and check that the call is presented. Do not take the call and wait for the call to be presented on the third phone. Take the call on the third phone. Check also the call can be picked up before the call is forwarded.	☒	☐	☐	No local Options in YEALINK SIP Phones to enable both the conditions at once.
5	**Phone puts call on-hold.** The phone is in conversation with another phone. This conversation is put on-hold. Check the display and on-hold music on this phone. Check also the display and audio signalization on The phone. Check the conversation can be retrieved.	☐	☒	☐	The LED blinks as the call is put on hold and the display also shows hold information.
6	**Phone is put on-hold.** The phone is in conversation with another phone. The other phone puts this conversation on-hold. Check the display and on-hold music on The phone. Check the conversation can be retrieved.	☐	☒	☐	On-hold Music heard and in the display also shows held information.
7	**Broker call.** The phone has two active conversations and switches from	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
	one to another. Check the display and on-hold music on the phones. Check also the display and audio signalization on the phone.				
8	**Transfer in conversation.** The phone has two active conversations and transfers the first to the second. Check the new conversation between the two other phones is successful and also the phone display (signalization of the transfer and back to idle state).	☐	☒	☐	
9	**Transfer during ringing.** The phone has one active conversation and another one in ringing step. Before the second callee takes the call, The phone transfers its first call to this second callee. Check the new conversation between the two other phones is successful and also The phone display (signalization of the transfer and back to idle state).	☐	☒	☐	
10	**Conference.** The phone has two active conversations and initiates a conference. Check the new conversation between the three parties is successful (audio and signalization).	☐	☒	☐	
11	**Do Not Disturb.** On The phone the local feature (if exists) Do not Disturb is activated. When calling this set, the call is not presented on the phone. On The phone the Do not Disturb is deactivated. When calling this set, the call is presented on the phone and can be picked up.	☐	☒	☐	We can see this as missed call in the display. But the phone in DND is not ringing.
12	**Wake Up.** On The phone the local feature (if exists) Wake Up is activated. When the wake up time arrives, the phone rings. When the picked up, the voice guide is played. Test also with The phone already in conversation when the wakeup time arrives. On The phone the Wake Up is activated then deactivated. When the previous wake up time arrives, nothing appends on the phone set.	☒	☐	☐	No local feature available in YEALINK SIP phones.

6.1.6 Other features

These tests check the phone behavior while using features like STP (date and time display).

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Date and time display using Network Time Protocol.** The SIP phone is configured to retrieve the date and time from a NTP server. Check the phone retrieves the information and displays it.	☐	☒	☐	

6.2 Hotel / Hospital tests

These tests check the phones behavior for SIP phone specific Hotel/Hospital features like provisioning, check-in and out, do not disturb, wake-up, room state modification, mini bar, forward, auto assignation and calls.

6.2.1 Provisioning

These tests check the phone provisioning as a room, suite, administrative or booth set.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Declare the SIP phone as a hotel room set** In the OXE configuration, the SIP set is declared as a hotel room set Or The SIP set is configured (user and password) with the parameters of an already declared SIP hotel room set The phone registers correctly to the OXE.	☐	☒	☐	Configured it as "Room & Normal"
2	**Declare the SIP phone as a hotel administrative set** In the OXE configuration, the SIP set is declared as a hotel administrative set Or The SIP set is configured (user and password) with the parameters of an already declared SIP hotel administrative set The phone registers correctly to the OXE.	☐	☒	☐	Configured it as "Administrative & Normal"
3	**Declare the SIP phone as a hotel booth set** In the OXE configuration, the SIP set is declared as a hotel booth set Or The SIP set is configured (user and password) with the parameters of an already declared SIP hotel booth set The phone registers correctly to the OXE.	☐	☒	☐	Configured it as "Home & Normal"

6.2.2 Check-in and check-out

These tests check the phone behavior after a check-in and a check-out as normal or VIP guest.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**A client checks in a room containing the SIP phone** In the OXE hotel menu (hotmenu), a check-in is done and the client gets the SIP phone room Or The SIP set is configured (user and password) with the parameters of an already declared SIP hotel room set in which a client has already checked-in.	☐	☒	☐	
2	**A client checks in as VIP in a room containing the SIP phone** Same as above but with the VIP parameter set. When this phone calls a hotel administrative set, the name displayed is completed with specific information.	☐	☒	☐	
3	**A client checks out from a room containing the SIP phone** In the OXE hotel menu (hotmenu), a check-out is done for the client using the SIP phone room	☐	☒	☐	

6.2.3 Do not disturb

These tests check the phone behavior in case of "Do not disturb" activation / deactivation (on the phone itself or from an administrative phone).

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Do Not Disturb is activated on the room SIP phone** On the room SIP phone the Do not Disturb is activated thanks to the prefix 42 (and then personal password) When calling this set, the call is not presented on the phone.	☐	☒	☐	
2	**Do Not Disturb is deactivated on the room SIP phone** On the room SIP phone the Do not Disturb is deactivated thanks to the prefix 42 When calling this set, the call is presented on the phone and can be picked up.	☐	☒	☐	
3	**Do Not Disturb is activated on the suite master SIP phone** On the suite master SIP phone the Do not Disturb is activated thanks to the prefix 586 When calling this guest (call to the guest number and call to the main and slave suite phones), the call is not presented on the phones. Same behavior when calling a slave SIP phone set.	☐	☒	☐	
4	**Do Not Disturb is deactivated on the suite master SIP phone** On the suite master SIP phone the Do not Disturb is deactivated thanks to the prefix 586 When calling this guest, the call is presented on the main suite phone and can be picked up. Same behavior when calling a slave SIP phone set.	☐	☒	☐	

6.2.4 Forward

These tests check the phone behavior in case of "Forward" activation / deactivation (on the phone itself or from an administrative phone), to an administrative set or to the voice mail. To allow the available forward functions it is required to have a checked-in guest with a voicemail.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Immediate forward is activated on the room SIP phone** On the room SIP phone the Forward is activated thanks to the prefix 51 (immediate forward). Forward destination is an administrative SIP set or the voice mail. When calling this set, the call is not presented on the phone but forwarded to the administrative phone or voice mail.	☐	☒	☐	Call is forwarded only when the GUEST is allocated with a voicemail.
2	**Forward is deactivated on the room SIP phone** On the room SIP phone the Forward is deactivated thanks to the prefix 41. When calling this set, the call is presented on the phone and can be picked up.	☐	☒	☐	
3	**Call to a phone, which is forwarded to another phone.** The phone calls a phone forwarded to a third phone. The third phone takes the call and the conversation is established.	☐	☒	☐	
4	**Forward on no answer to another phone. Prefix: 53** The phone is forwarded on no answer to another phone. Call the phone and check that the call is presented on analog phone. Do not take the call and wait for the call to be presented on the third phone. Take the call on the third phone. Check also the call can be picked up before the call is forwarded.	☐	☒	☐	Call is forwarded only when the GUEST is allocated with voicemail.
5	**Forward on busy to another phone. Prefix: 52** The phone is forwarded on busy to another phone. While the phone is already in conversation, call The phone and check that the call is presented. Do not take the call and wait for the call to be presented on the third phone. Take the call on the third phone.	☐	☐	☒	OXE prefix not supported in hotel configuration.
6	**Forward on busy / no answer to another phone. Prefix: 54** The phone is forwarded on busy / no answer to another phone. While the phone is already in conversation, call The phone and check that the call is presented. Do not take the call and wait for the call to be presented on the third phone. Take the call on the third phone. Call The phone and check that the call is presented. Do not take the call and wait for the call to be presented on the third phone. Take the call on the third phone. Check also the call can be picked up before the call is forwarded.	☐	☐	☒	OXE prefix not supported in hotel configuration.

6.2.5 Voice mail

These tests check the phone behavior when interworking with the OXE 4645 voicemail.
In order to do these tests, the Room has to be checked in. If the room isn't checked-in there is no voicemail available. In the User parameters, make sure that before the Check-in there is no Voicemail number.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Voice mail message signalization**. The voice mail (OXE 4645) number is configured in the phone. Call The phone and leave a message to its voice mail (for example by forwarding the phone to the voice mail). Check that the message is indicated on the phone (led or display).	☐	☒	☐	In the display of the Yealink phone, the number of New & Old messages are one at the same (always the number of new messages), irrespective of the actual number of old messages.
2	**Voice mail message listening.** The phone has a voice mail message (see above). Press the voice mail key and interacts with the voice mail to listen to the message. Check the led or display does not show any new message as soon as the last one is read.	☐	☒	☐	
3	**Voice mail message deposit.** The phone calls another phone forwarded to the voice mail. He leaves a message. Check the interaction between the phone and voice mail. Listen to the message from the other phone.	☐	☒	☐	
4	**Voice mail message deposit. Suffix: 8** Check the can leave a voice message to a phone which is not answering or already in conversation thanks to OXE suffix.	☐	☐	☒	From YEALINK SIP phone we are unable to leave messages to other user by dialing the suffix 8. Even from IPTouch we can only leave the message by pressing the soft key in the display when we dial busy user, and not by dialing the suffix 8. SR: 1-118746681

6.2.6 Wake up

These tests check the phone behavior in case of "Wake up" activation / deactivation (on the phone itself or from an administrative phone). To set a wake-up call it is necessary to dial the extension mentioned in the test case and then after hearing the tone dial: “HH MM XXXX”. With “HH MM” being the time of the alarm call and “XXXX” being the room extension number or the guest number. After setting up the alarm call an audio message should confirm the time set. To cancel an alarm call, dial the necessary extension mentioned in the test case and after hearing the tone dial the extension aimed by the cancellation.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Wake Up is activated on the room SIP phone** On the room SIP phone the Wake Up is activated thanks to the prefix 506 When the wake up time arrives, the phone rings. When the picked up, the voice guide is played.	☐	☒	☐	
2	**Wake Up is deactivated on the room SIP phone** On the room SIP phone the Wake Up is deactivated thanks to the prefix 507 When the previous wake up time arrives, nothing appends on the phone set.	☐	☒	☐	
3	**Suite wake Up is activated on a suite SIP phone** On the suite master SIP phone the suite wake Up is activated thanks to the prefix 584 When the wake up time arrives, the entire suite phones are ringing. When picking up on one phone, the voice guide is played and all the other phones stop ringing. Test also the activation from slave suite phones. Behavior is the same.	☐	☒	☐	
4	**Room wake Up is activated on a suite SIP phone** On the suite master SIP phone the suite wake Up is activated thanks to the prefix 506 When the wake up time arrives, only the suite phone on which the wake up has been set is ringing. All the other suite phones do not. When picking up, the voice guide is played. Test also the activation from slave suite phones. Behavior is the same.	☐	☒	☐	
5	**Suite wake Up is deactivated on a suite SIP phone** On the suite master SIP phone the Wake Up is deactivated thanks to the prefix 585 When the previous wake up time arrives, nothing appends on the phone sets (master and slaves). Test also the deactivation from slave suite phones. Behavior is the same.	☐	☒	☐	

6.2.7 Change room state

These tests check the phone can change the room state (as a room or administrative set).

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**The room state is changed on the room SIP phone set.** On the room SIP phone the room state is changed thanks to the prefix 587 (then made personal code, then room status). Check the new room state thanks to the hotel menu (hotmenu). Several room states are tried (1 = done and available, 2 = to do completely, 3 = to do partially, 4 to 9 = problem).	☐	☒	☐	

6.2.8 Mini bar

These tests check the phone can change the "mini bar" state (as a room or administrative set).

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**The min-bar state is changed on the room SIP phone set.** On the room SIP phone the mini-bar state is changed thanks to the prefix 588 Several mini-bar states are tried. This test can be validated when analyzing the hotel traces on the OXE. The only verification done is to see if the correct digits are sent to the OXE.	☐	☒	☐	

6.2.9 Calls

These tests check the phone behavior during calls between rooms, from and to an administrative set, external outgoing call. It also checks the transfer feature.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**SIP room phone set to another SIP room phone set (different rooms).** Using guest and room numbers.	☐	☒	☐	
2	**SIP room phone to an administrative phone set and vice versa.** Using guest and room numbers.	☐	☒	☐	
3	**SIP room phone set to an external number (T0/T2 for example) and vice versa**	☐	☒	☐	
4	**SIP suite phone set to an external number (T0/T2 for example) and vice versa.** Try with the master and several slave phone sets.	☐	☒	☐	
5	**SIP booth phone set to an external number (T0/T2 for example) and vice versa.**	☐	☒	☐	
6	**SIP room phone to an attendant and vice versa.** Call is properly established.	☐	☒	☐	
7	**Incoming external call (T0/T2 for example) to an attendant phone set which transfers the call to the SIP room phone. Transfer is done while the SIP phone is ringing but also after this one has picked up the call (using the attendant soft key or going on hook).** Call is properly established.	☐	☒	☐	
8	**Outgoing call from a SIP phone to an attendant with transfers to an external call (T0/T2 for example).** Call is properly established.	☐	☒	☐	
9	**Call back on no answer / busy. Suffix: 5** The phone calls another phone already in conversation (or does not answer – check both). The phone set uses the call back suffix to be recalled. Check the call back query is taken into account and processed.	☐	☒	☐	
10	**Conference. Suffix: 3** Check the phone can establish a three party conference thanks to OXE suffix.	☐	☐	☒	OXE suffix is not supported on SIP phones in Hotel Mode. Has to use local feature.
11	**Broker call. Suffix: 1** Check the phone can switch between two calls thanks to OXE suffix.	☐	☐	☒	OXE suffix is not supported on SIP phones in Hotel Mode. Has to use local feature.
12	**Enquiry call. Suffix: 2** Check the phone can make a second call while already in conversation thanks to OXE suffix.	☐	☐	☒	OXE suffix is not supported on SIP phones in Hotel Mode. Has to use local feature.

6.2.10 Multi line

These tests check the phone behavior during multi line calls from and to different kind of phone set types (SIP, IPTouch, UA). Room and suite phones are tested.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Incoming call to a room phone while already in conversation** The phone (only phone of the room) is already in conversation and receives a new incoming call. Check the display (new call presentation) and audio (new call signalization). Check this call can be taken (and that the first one is put on hold). Check the phone can switch between the two calls (and that the other parties are put on hold).	☐	☒	☐	Incoming call in the second line makes the LED blinks and held tone is heard when we are active in the first call.
2	**Outgoing call from a room phone while already in conversation** The phone (only phone in the room) is already in conversation and makes a new outgoing call. Check the display (new call presentation) and audio (new call signalization). Check this call can be taken (and that the first one is put on hold). Check the phone can switch between the two calls (and that the other parties are put on hold).	☐	☒	☐	
3	**Incoming call to a room phone while already in conversation on the two lines** The phone (only phone in the room) is already in conversation on its two lines and receives a new incoming call. Check the display (new call presentation) and audio (new call signalization).	☐	☐	☒	We can make the third call to wait by using the suffix, when we are active in two calls but unable to take the same after it finishes the two calls. No display and no held tone for this call but calling end have to wait and alerts once this two calls are disconnected. We can only make a single call to wait using the suffix in hotel mode.
4	**Incoming call to suite phone while already in conversation** The phone (main phone of a suite) is already in conversation and receives a new incoming call. Check the display (new call presentation) and audio (new call signalization). Check this call can be taken (and that the first one is put on hold). Check that the slave phones stop ringing. Check the phone can switch between the two calls (and that the other parties are put on hold).	☐	☒	☐	
5	**Outgoing call from a suite phone while already in conversation** The phone (main phone of a suite) is already in conversation and makes a new outgoing call. Check the display (new call presentation) and audio (new call signalization). Check this call can be taken (and that the first one is put on	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
	hold). Check the phone can switch between the two calls (and that the other parties are put on hold).				
6	**Incoming call to a suite phone while already in conversation on the two lines** The phone (main phone of a suite) is already in conversation on its two lines and receives a new incoming call. Check the display (new call presentation) and audio (new call signalization). Check that the slave phones ring and can take the call.	☐	☒	☐	
7	Repeat steps 4 to 6 but this time the SIP phone is a suite slave phone.	☐	☒	☐	
8	**Outgoing call from a room phone to another room phone which is already in conversation** The phone calls another room phone (only phone in the room) which is already in conversation and, thanks to Busy Camp On feature (suffix 6), enforce the called to phone to ring. Check the display (new call presentation) and audio (new call signalization). Check this call can be taken (and that the first one is put on hold). Check the phone can switch between the two calls (and that the other parties are put on hold).	☐	☒	☐	
9	Repeat previous steps but, this time, use guest number (and not room number) when calling to a room.	☐	☒	☐	

6.2.11 Multi occupation

These tests check the phone behavior when several guests are located in the same room.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**SIP set registration to OXE** The phone successfully registers to the OXE	☐	☒	☐	
2	**Incoming call to the room phone number** Another phone (IPTouch) calls the room phone number. The call can be picked up and is successfully established.	☐	☒	☐	
3	**Incoming call to the first guest phone number** Another phone (IPTouch) calls the first guest phone number. The call can be picked up and is successfully established.	☐	☒	☐	
4	**Incoming call to the second guest phone number** Another phone (IPTouch) calls the second guest phone number. The call can be picked up and is successfully established.	☐	☒	☐	
5	**Outgoing call by the first guest** In case of an external call (to a PSTN user for example), the first guest makes an outgoing call using his guest ID. The call is successfully established.	☐	☒	☐	
6	**Outgoing call by the second guest** In case of an external call (to a PSTN user for example), the second guest makes an outgoing call using his guest ID. The call is successfully established.	☐	☒	☐	

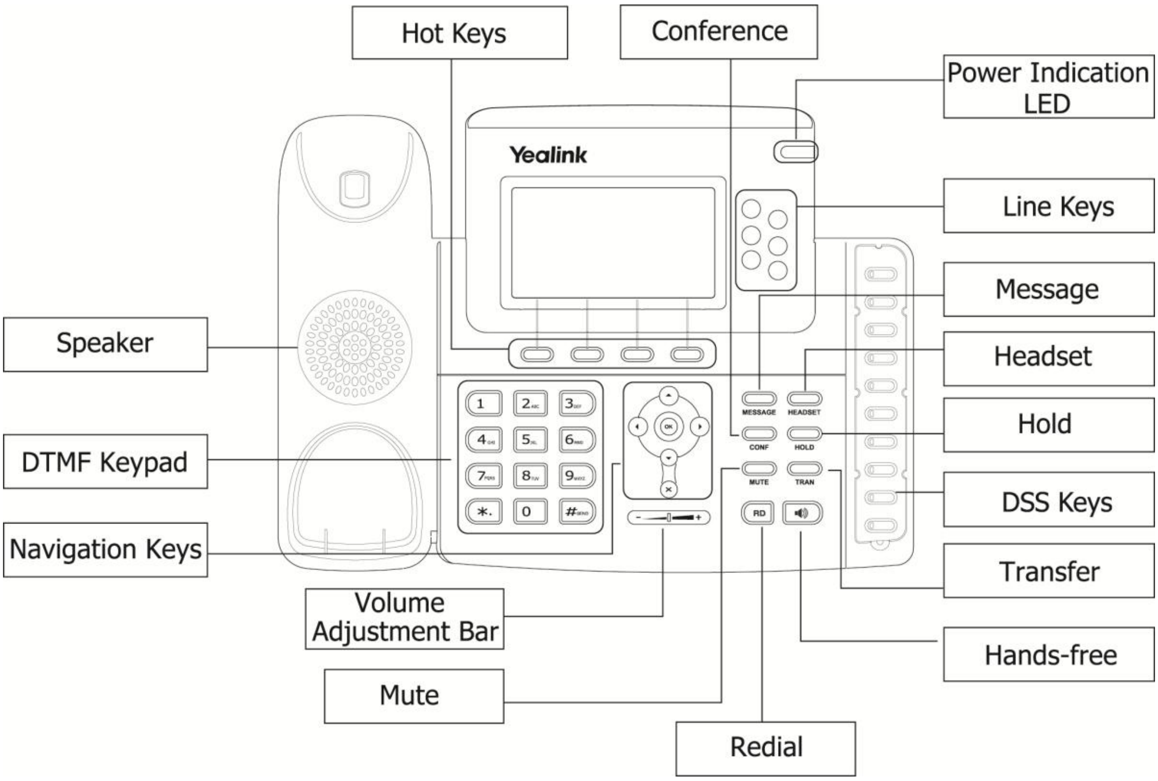
6.2.12 Additional Tests.

These tests check the good behavior in hotel suite mode.

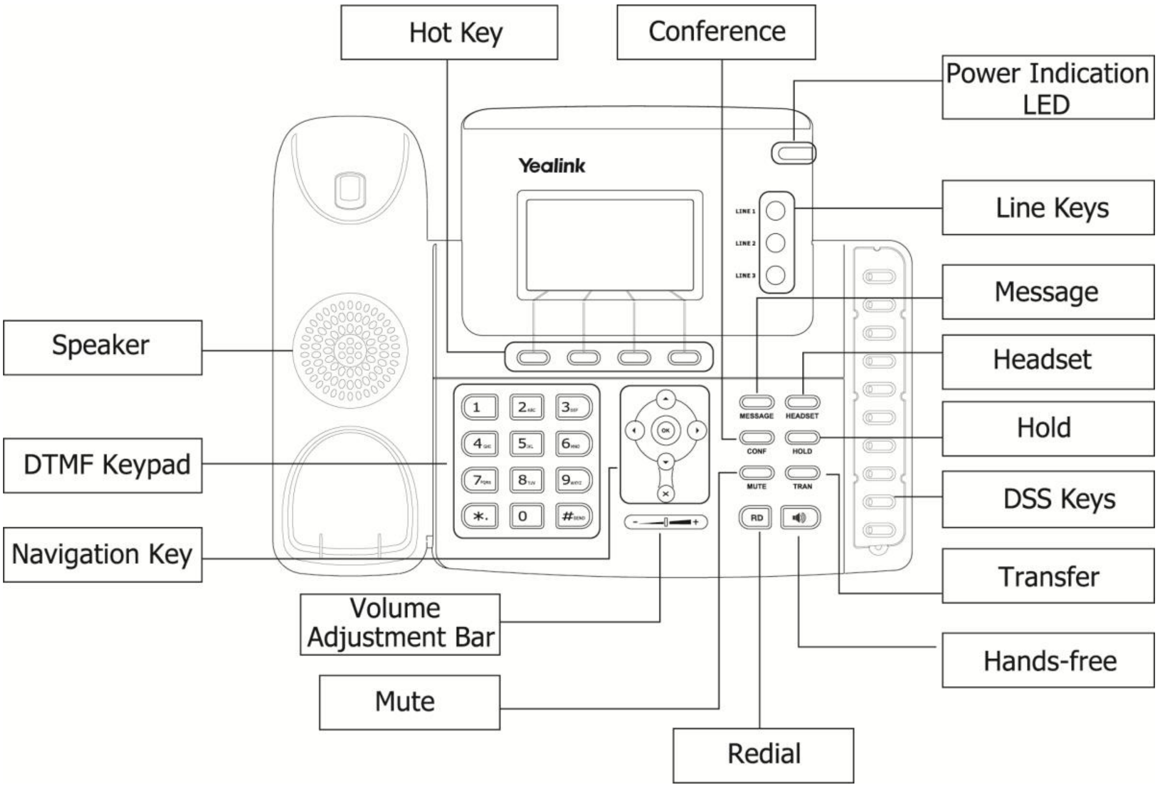
Test Case Id	Test Case	N/A	OK	NOK	Comment
1	For a suite with several phones, check behavior when there is an incoming call (to the suite phone number) while there is already a phone in conversation	☐	☒	☐	
2	Check that after a checkout all the guest specific information are erased (voice mail messages, phone state like forward, do not disturb, wake up time)	☐	☒	☐	
3	Test that DID attribution is working fine with the SIP hotel sets	☐	☒	☐	

7 Appendix A: Partner Application Description

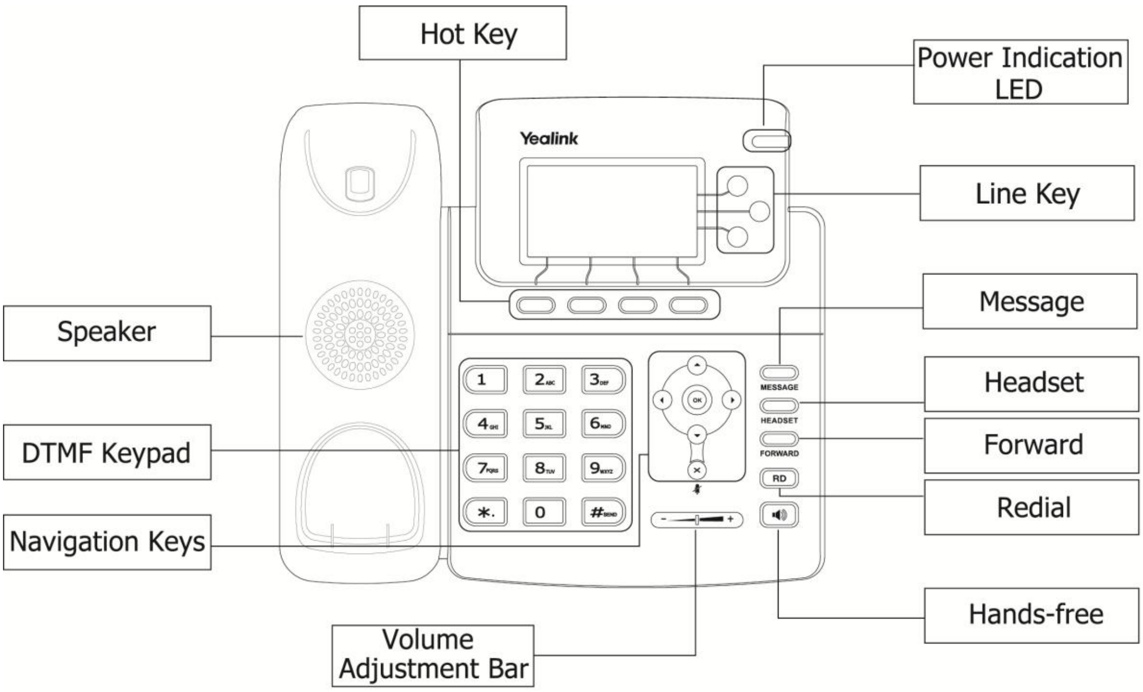
YEALINK T28P SIP Phone:



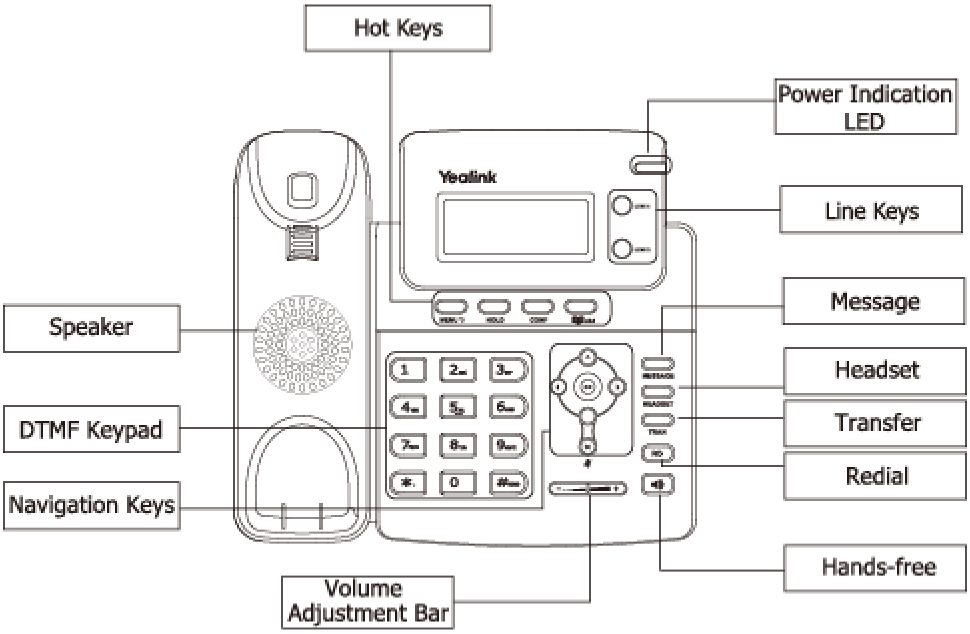
YEALINK T26P SIP Phone:



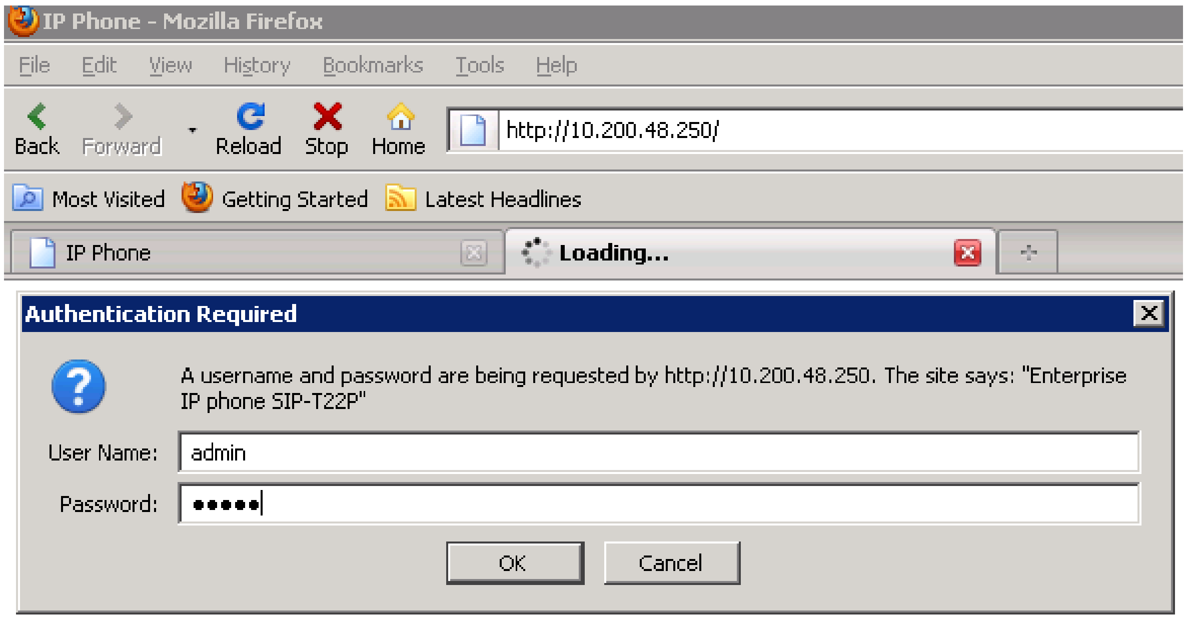
YEALINK T22P SIP Phone:



YEALINK T20P SIP Phone:



Webpage Login:



Access to the Admin Home page (Web interface)

1. Access your web browser. Enter the IP address on your browser. Example: http://10.200.48.250 (Phone IP Address).

2. The Web login page will be displayed. Enter the user name and the password and click **Login**. The administrator's default user name and password are "**admin**" and "**admin**" respectively.

8 Appendix B: Partner Application: Configuration Requirements

Phone Page Version Information:

Version	
Firmware Version	7.60.0.120
Hardware Version	5.0.0.53
Network	
WAN Port Type	Static IP
WAN IP Address	10.200.48.250
Subnet Mask	255.255.255.0
MAC Address	00-15-65-24-EC-56
Link Status	Connected
PC IP Address	0.0.0.0
Device Type	Bridge
DHCP Server Status(PC)	Disabled

SIP Page configuration:

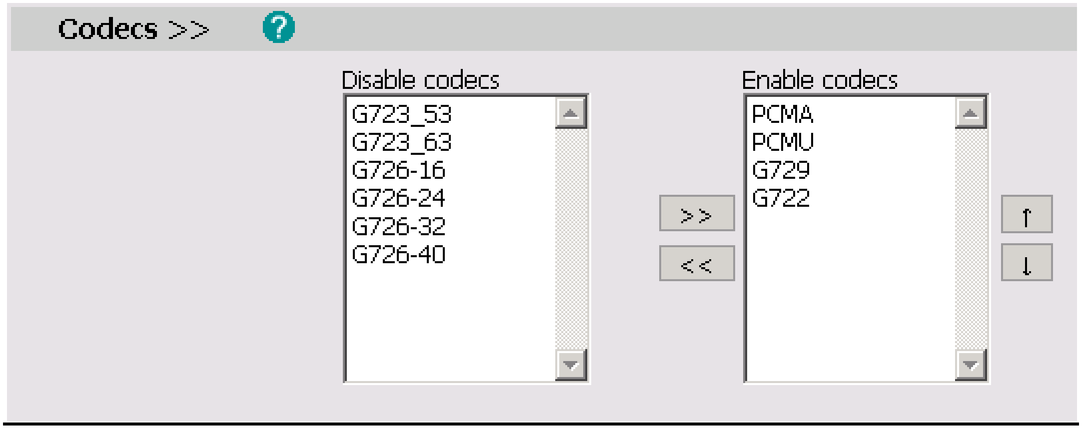
Login to the webpage and Go to SIP Account Settings for the SIP account.

Account	Account 1	
Basic >>		
Register Status	Registered	
Account Active	On / Off (On selected)	
Label	1240	
Display Name	1240	
Register Name	1240	
User Name	1240	
Password	●●●●●●●●●	
SIP Server	node1slash.etesting.c	Port 5060
Enable Outbound Proxy Server	Enabled	
Outbound Proxy Server	node1slash.etesting.c	Port 5060
Transport	UDP	
Backup Outbound Proxy Server	node1slash.etesting.c	Port 5060

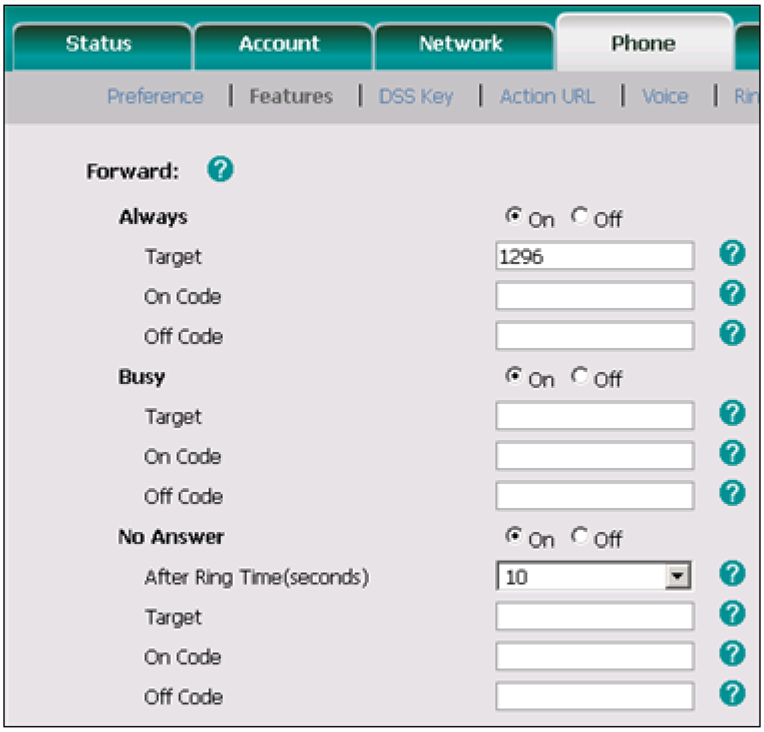
A SIP account must be created. There are two SIP Profile supported in the VTECH two line models. You can select any one line and load the same to configure the SIP account. Enter your Extension, Authentication Name and Password as mentioned above. Make sure you activate the account by selecting the line, configure and then click on "SAVE" at the bottom after configured the settings.

Codec Configuration

We can configure the order of Codec should be used in the Audio Codec Settings Tab.



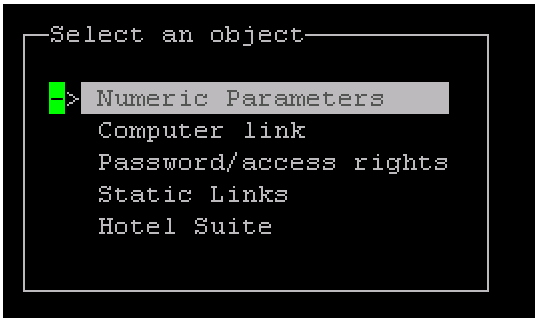
Phone Tab Configuration

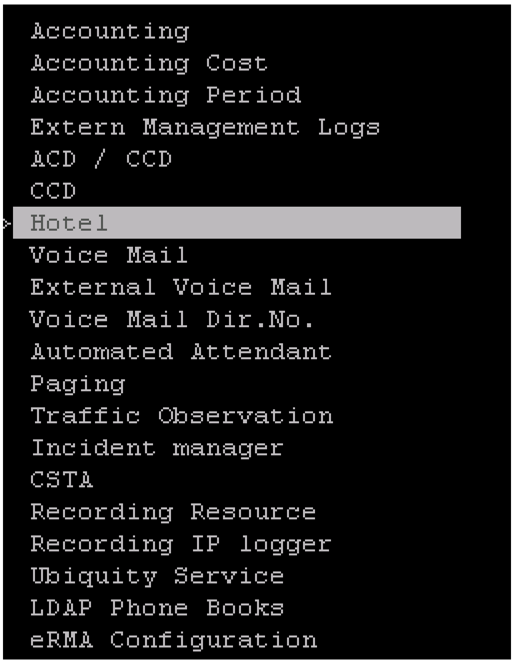


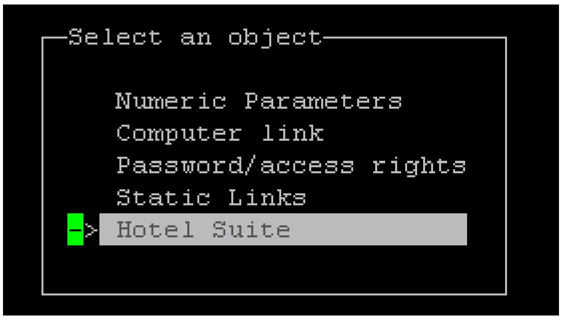
9 Appendix C: Alcatel-Lucent Platform: configuration requirements

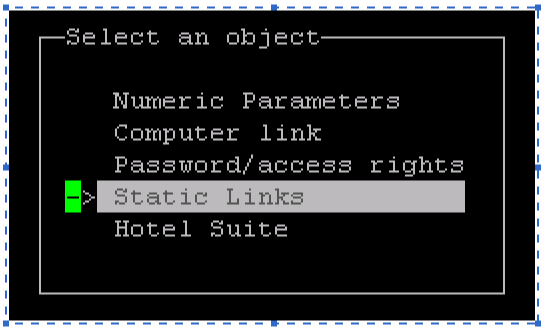
Access to Hotel application: to configure the hotel application parameters.

.









Creation of a room SIP phone set: add a SIP extension user.

```
Review/Modify: Users

            Node Number (reserved) : 101
                  Directory Number : 1239

                    Directory name : Yealink T22P
              Directory First Name : --------------------
              UTF-8 Directory Name : ---------------------------------------
        UTF-8 Directory First Name : ---------------------------------------
                     Location Node : 1
                     Shelf Address : 255
                     Board Address : 255
                 Equipment Address : 255
                          Set Type + SIP extension
                     Entity Number : 1
                      Set Function : Default
```

Configuration of the used IP addresses of a room SIP phone set: in order to be set in the correct OXE IP domain (mgr / users / IP SIP Extension).

```
Review/Modify: IP SIP Extension

            Node Number (reserved) : 101
                  Directory Number : 1239
                  Directory Number : 1239

                          Set Type + SIP extension
                        IP Address : 10.200.48.249
```

Disabling the protection against beeps for the room SIP phone set: This allows the presentation of a second incoming call.

```
Review/Modify: Phone Features COS

           Routing Mode At Off-hook + NO Routing
        Inter-Company Calling Right + False

                                      Set features

                  Immediate forward : 1
          Immediate forward on busy : 1
               Forward on no answer : 1
       Forward on busy or no answer : 1
               Forward cancellation : 1
        Forward cancel.by destinat. : 0
   Overfl.on no answer to associate : 1
         Cancel Overfl.to associate : 1
                    Sta. group exit : 0
                   Sta. group entry : 0
  Protect. against barge-in & beeps : 0
                               Lock : 1
                    Auto-assignment : 0
                       Substitution : 1
              Password modification : 1
```

Creation of a hotel guest: add a guest user (**choose 255 for Cristal, Board and Equipment address**).

```
      Node Number (reserved) : 101
            Directory Number : 1277

              Directory name : TGUEST 7
        Directory First Name : --------------------
         UTF-8 Directory Name : ----------------------
  UTF-8 Directory First Name : ----------------------
               Location Node : 1
               Shelf Address : 255
               Board Address : 255
           Equipment Address : 255
                    Set Type + ANALOG
               Entity Number : 1
                Set Function + Default
                Profile Name : ---------------
                Key Profiles + None
           Domain Identifier : 0
                 Language ID : 1

                 Secret Code : ----
```

Creation of an administrative SIP phone set: add a SIP extension user.

```
      Node Number (reserved) : 1
            Directory Number : 1282

              Directory name : AdminSIP
        Directory First Name : --------------------
         UTF-8 Directory Name : ------------------------------
   UTF-8 Directory First Name : ------------------------------
               Shelf Address : 255
               Board Address : 255
           Equipment Address : 255
                    Set Type + SIP extension
               Entity Number : 1
                Set Function + Default
                Profile Name : ---------------
                Key Profiles + None
           Domain Identifier : 0
                 Language ID : 1

                 Secret Code : ****
                     Confirm : ****
```

Creation of the trunk group used for the local SIP: add a trunk group under mgr / Trunk groups.

```
            Node Number (reserved) : 101
                    Trunk Group ID : 100
               Instance (reserved) : 1

                  Trunk Group Type + T2
                  T2 Specification + SIP
               Public Network Ref. : ------
           VG for non-existent No. + YES
                     Entity Number : 0
             Supervised by Routing + NO
      VPN Cost Limit for Incom.Calls : 0
 Immediate Trk Listening if VPNCall + YES
                          VPN TS % : 50
                    CSTA-Monitored + NO
```

Configuration of the SIP gateway: under mgr / SIP

```
      Node Number (reserved) : 101
         Instance (reserved) : 1
         Instance (reserved) : 1

              SIP Subnetwork : 10
             SIP Trunk Group : 100
                  IP Address : 10.10.10.50
         Machine name - Host : node1slash
       SIP Proxy Port Number : 5060
  SIP Subscribe Min Duration : 15
  SIP Subscribe Max Duration : 5060
               Session Timer : 300
           Min Session Timer : 100
        Session Timer Method + UPDATE
       DNS local domain name : etesting.com
                    DNS type + DNS SRV
         SIP DNS1 IP Address : 10.1.8.1
         SIP DNS2 IP Address : 10.10.10.50
                  SDP in 18x + True
                 Cac SIP-SIP + False
```

Configuration of the SIP proxy: under mgr / SIP

```
             Node Number (reserved) : 101
                Instance (reserved) : 1
                Instance (reserved) : 1

               SIP initial time-out : 500
                      SIP timer T2 : 4000
                Dns Timer overflow : 5000
                  Recursive search + True
      Minimal authentication method + SIP None
              Authentication realm : ----------------
Only authenticated incoming calls + False
                  Framework Period : 3
    Framework Nb Message By Period : 25
       Framework Quarantine Period : 1800
           TCP when  long messages + True
```

Configuration of the hotel parameters: under mgr / Application / Hotel

```
No. Of Secret Code Errors : 0
     Disabled Code Period : 0
          Management mode + Guest management
         Night Audit Time + No Night Audit
             Installation + Hotel
```

Configuration of the multi occupation type: under mgr / Application / Hotel / Numerical parameters

```
   Node Number (reserved) : 101
      Instance (reserved) : 1
      Instance (reserved) : 1
                Parameter + Multiple Occupancy Type

 Multiple Occupancy Type + Static Multiple Occupancy
```

Configuration of the suite wake up type: under mgr / Application / Hotel / Numerical parameters

```
  Node Number (reserved) : 101
     Instance (reserved) : 1
     Instance (reserved) : 1
               Parameter + Suite Wake-up Type

     Suite Wake-up Type + Wake-up for the single set only
```

Configuration of a suite: under mgr / Application / Hotel / Hotel suite

```
          Node Number (reserved) : 1
             Instance (reserved) : 1
             Instance (reserved) : 1
              Main Directory No. : 1299

                 Affected room Directory No.

    [  Add   ]    [ Remove ]    [  Next  ]    [Previous]

  Affected room Directory No. : 1294
```

Access to the hotel application menu: use 'hotmenu' command

```
:------------------------------------------------------------------------------:
: 1  : Check-in                        : 2  : Check-out                        :
:------------------------------------------------------------------------------:
: 3  : Itemized billing                : 4  : Repertory                        :
:------------------------------------------------------------------------------:
: 5  : Billing                         : 6  : Wake-up                          :
:------------------------------------------------------------------------------:
: 7  : Message                         : 8  : Call Redirection                 :
:------------------------------------------------------------------------------:
: 9  : Room status                     : 10 : Inter-room calls                 :
:------------------------------------------------------------------------------:
: 11 : Assign-Change room             : 12 : Utilities                        :
:------------------------------------------------------------------------------:
: 13 : Night-Audit                     : 14 : Suites                           :
:------------------------------------------------------------------------------:
```

10 Appendix D: Partner escalation process

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

11 Appendix E: AAPP program

11.1 Alcatel-Lucent Application Partner Program (AAPP)

Complete e-business solutions at your disposal

The Application Partner Program is designed to support companies that develop communication applications for the enterprise market, based on Alcatel-Lucent's Omni product family.

The program provides tools and support for developing, verifying and promoting compliant third-party applications that complement Alcatel-Lucent's Omni-based products. 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, …

Web site

If registered Application Partner, you can access the AAPP website at this URL:
http://applicationpartner.alcatel-lucent.com

11.2 Alcatel-Lucent.com

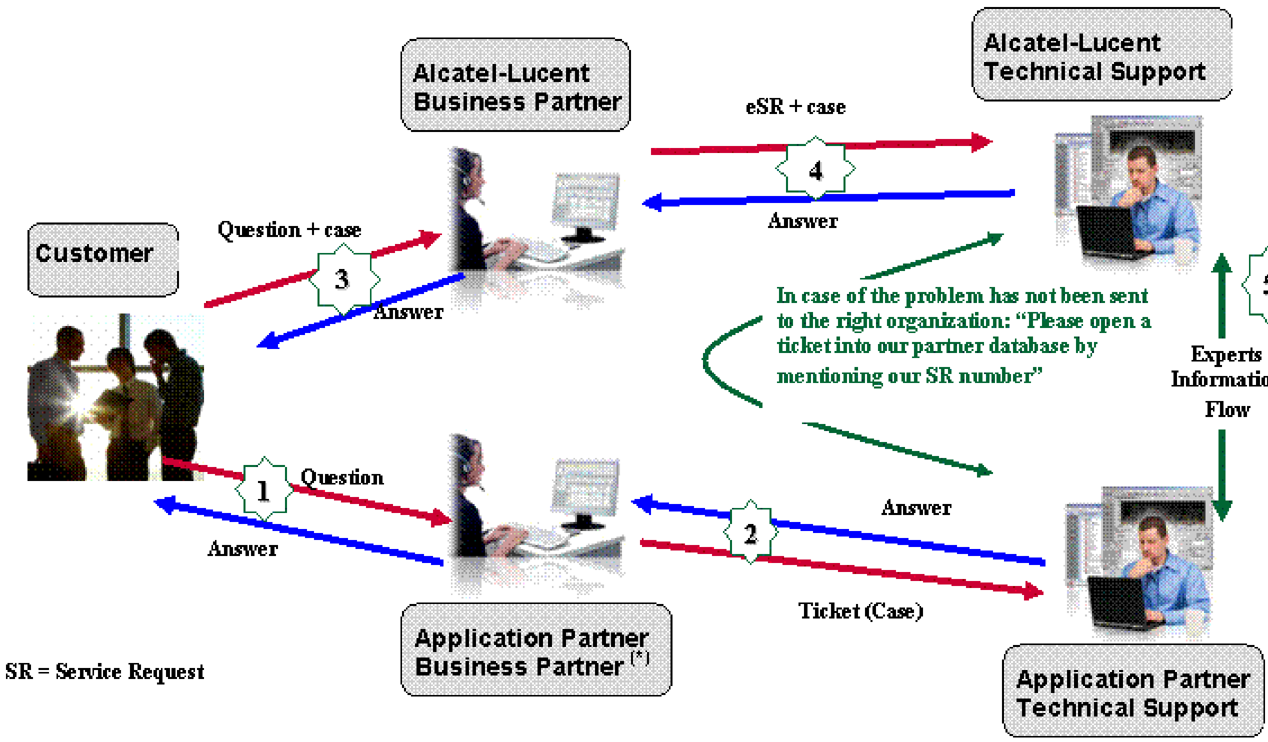
You can access the Alcatel-Lucent website at this URL: http://www.Alcatel-Lucent.com/

12 Appendix F: AAPP Escalation process

12.1 Introduction

The purpose of this appendix is to define the split of responsibilities and the escalation process to be applied by the Alcatel-Lucent Business Partners when facing a problem with a solution involving an Alcatel-Lucent platform and a Third-Party application *with or without a valid Alcatel-Lucent Inter-Working Report.*

If a problem occurs on an installation involving Alcatel-Lucent platforms and a certified product or application, both parties, Alcatel-Lucent and the Application Partner, are engaged as follows:



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

12.2 Escalation in case of certified application/products

The Alcatel-Lucent support will be limited to applications with a valid Inter-Working Report (IWR). Known problems or remarks mentioned in the IWR will not be taken into account.

A valid IWR means an official IWR exists which is posted on the Alcatel-Lucent Enterprise Business Portal and mentions the same release/version of the software of both parties as those of the current customer installation (Or an official agreement between Alcatel-Lucent and the Third-Party exists to support the customer installation if the release/version doesn't match those mentioned in the latest IWR).

If there is an interworking issue, 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.

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 a guarantee of the availability of the solution. 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.

12.3 Escalation in case of non-certified application/product

If an Alcatel-Lucent Business Partner escalates an issue where a 3rd party application is involved and the following conditions apply:

1. no IWR exist (not available on the Enterprise Business Portal for Business Partners or on the Alcatel-Lucent Application Partner web site) ,

2. Or the 3rd party company is referenced as AAPP participant but with no existing IWR,

3. Or the existing IWR is available but the release/version of the both parties (Alcatel-Lucent and 3rd-party) are not the same than those currently deployed at the customer site (see exception in Note 2).

In this case, the only responsibility of the Alcatel-Lucent Technical Support is to verify that the Alcatel-Lucent platform is correctly installed and configured for a standard use and that the Alcatel-Lucent equipments perform as expected. If that's the case, Alcatel-Lucent will be forced to close the case.

If the Alcatel-Lucent Business Partner, the customer or the 3rd party company need additional and specific involvement from Alcatel-Lucent, there are two options:

- Either request a quote for specific investigation and diagnosis (with no agreement to fix the issue),
- Or the AAPP program process is followed to officially certify the 3rd party application/product.

For both options, just send the request to the AAPP team (by opening an e-SR).

IMPORTANT NOTE 1: Even if the 3rd party company is able to demonstrate the issue is on the Alcatel-Lucent side, there is no obligation from Alcatel-Lucent to fix it (there is no official IWR established between the two parties).

IMPORTANT NOTE 2: For case 3, Alcatel-Lucent and the Third-Party company may decide to provide a document specifying the possible extension of the IWR by mentioning the list of releases/versions officially supported. (Another way is to update an existing IWR with new release/version compatibility).

12.4 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:

<table>
<tr><th>Country</th><th>Supported language</th><th>Toll free number</th></tr>
<tr><td>France</td><td rowspan="3">French</td><td rowspan="23">+800-00200100</td></tr>
<tr><td>Belgium</td></tr>
<tr><td>Luxembourg</td></tr>
<tr><td>Germany</td><td rowspan="3">German</td></tr>
<tr><td>Austria</td></tr>
<tr><td>Switzerland</td></tr>
<tr><td>United Kingdom</td><td rowspan="16">English</td></tr>
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<tr><td>Greece</td></tr>
<tr><td>Slovakia</td></tr>
<tr><td>Portugal</td></tr>
<tr><td>Spain</td><td>Spanish</td></tr>
</table>

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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