Alcatel·Lucent
Certified Application

Alcatel Lucent Application Partner Program Inter-Working Report

Partner: Technicolor
Application type: VoIP SIP HOTEL Phone
Application name: SIP TB30
Alcatel-Lucent Platform: OmniPCX Office

technicolor

Tests identification

Date of the tests	February 2012

Alcatel-Lucent's representative	Alain Botti
Partner's representative	Wim Sohier

Alcatel-Lucent Communication Platform (OmniPCX 4400/Enterprise, OmniTouch, OmniPCX Office, ...)	OmniPCX Office
Alcatel-Lucent compatibility release	R810 / 045.003
Partner's application version	TB30 V1.70.4

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Test results

☐ Passed ☐ Refused ☐ Postponed

☑ Passed with restrictions

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

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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 verify 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.

Note: This interworking report does not cover configuration/management and/or mass provisioning the TB30 SIP Set. For any questions related to these topics, please contact Technicolor.

2 Application information

Application type: VoIP SIP Phone

Application commercial name: SIP Phone Model TB30

Application version: TB30 Firmware version V1.70.4

Interface type : SIP/Ethernet

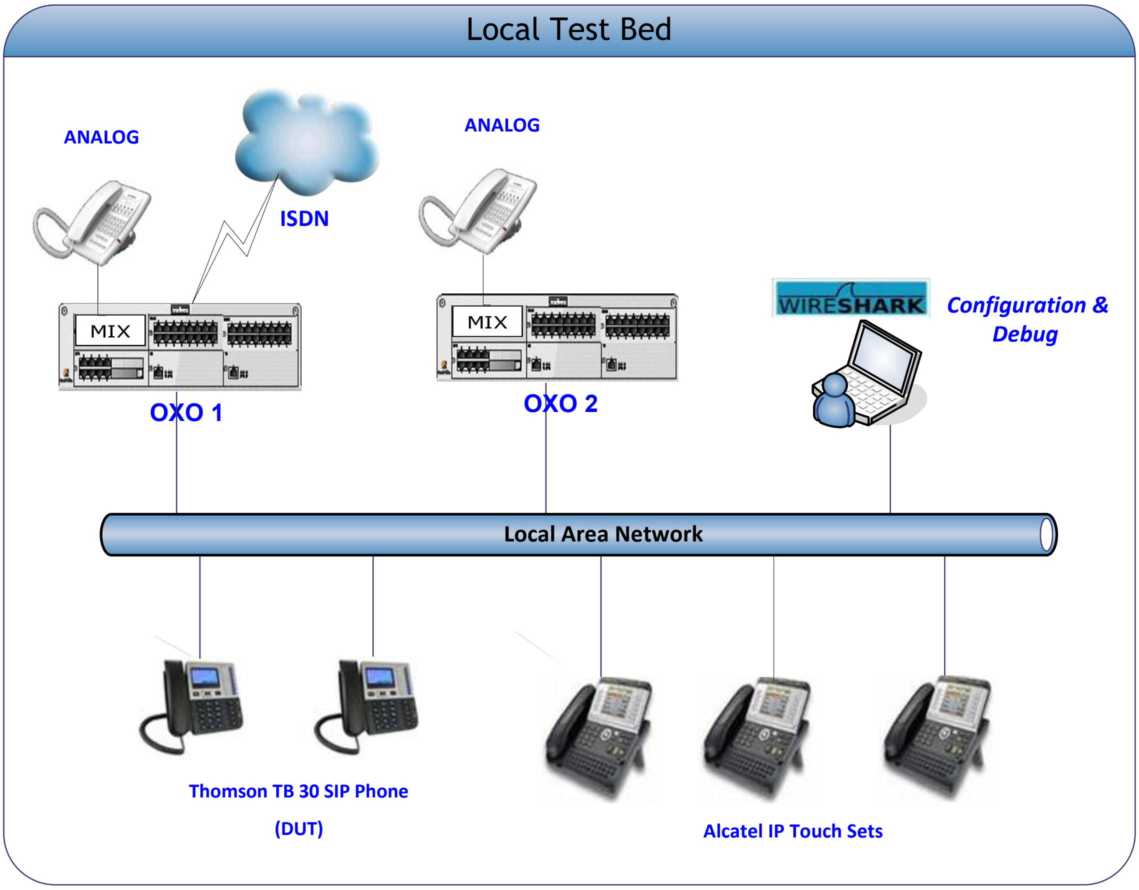
Interface version (if relevant):

Brief application description:

Technicolor's ranges of corded and cordless phones have all the latest features for residential and professional communications. Improving communication with style and advanced technology, Technicolor works hand-in-hand with broadband operators in the development of customized telephone handset products with value-adding features to support their advanced service offerings like phone mail, caller ID, and more.

Type of application/product:

3 Test Environment



3.1 Hardware configuration

Alcatel-Lucent Communication Platform:

- OmniPCX Office Rack
- PowerCPU
- Release: R810/045.003
- OMC: R810/22.1a

Setup Details:

Setup Information OXO 1	
OXO 1 IP address	10.130.164.64
Domain name	Oxoone.testandvalidate.com
Voicemail No	500
Attendant No	300
OXO Extension Details used for test	
IP Touch numbers	322, 323 & 324
TB30 Dir numbers	397, 398 and 399
UA Set No	301

Setup Information OXO 2	
Network OXO address	10.130.158.88
Network OXO Domain name	Oxotwo.testandvalidate.com
Network OXO Extension Details used for test	
TB30 Dir numbers	122 & 123
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.

3.2 Software configuration

- **Alcatel-Lucent Communication Platform**: OmniPCX Office R810/045.003
- **Partner Application**: TB30 V1.70.4

Note: TB30 is registered in the OmniPCX Office as "Open SIP phone".

4 Summary of test results

4.1 Summary of main functions supported

Features	Status	Comments
Initialization including network configuration	OK	
SIP registration	OK	DHCP registration is not supported for SIP phones
SIP authentication	OK	
Voice over IP and RTP codec support	OK	
Outgoing Call	OK_But	Display is not updated when connected line is not the called party
Incoming Call	OK_But	Calls with Name containing special characters may fail
Features During Conversation	OK	Only available from device (local)
Call Transfer	OK	Semi – Attended / Unattended (Blind)Transfer is not supported for SIP sets on OXO R8.1
Attendant	OK	
Voice mail interaction and indication	OK	

4.2 Summary of problems

- Call forward activated in call server via prefix is working but wrong acknowledgement message is send to SIP device - crms00351769.
- Call forward on busy is not working for SIP phones.
- Appointment configuration in call server is not working.
- When TB30 SIP set is calling the forwarded user, the SIP set display is not updated with the redirected user information after the call established.

4.3 Summary of limitations

- DHCP mode 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.
- SIP password is hidden in the webpage (******) but is visible in clear text on the TB30 display in MMI admin session.
- No count of new messages (voice mails) available on the display of TB30.
- Call feature activation in the call server (eg CFU/CFB) is not displayed on the SIP device.
- Semi - Attended and blind Transfer are not supported in OXO

4.4 Notes, remarks

- TB 30 is registered in the OmniPCX Office as "Open SIP phone".

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 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.

6 Test Results

6.1 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 397, 398 & 399 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 397, 398 & 399 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.	☐	☒	☐	
3	**DHCP registration (with OXO internal DHCP server)**	☒	☐	☐	Restricted to ALU IP extensions
4	**NTP registration** The SIP phone 399 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 398 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 398 to use the protocol SIP over UDP and other TCP In the two cases, check the registration and basic calls.	☐	☒	☐	

6.2 Audio codec negotiations/ VAD / Framing

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

Phone configuration: configure TB30 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 TB 30 Call from SIP 398 to IP Touch 323 Check that the call is established in G711 A-law. Check audio quality Call from IP Touch 323 to SIP 398 Check that the call is established in G711 A-law. Check audio quality	☐	☒	☐	
2	Select G729 as 1st codec in TB 30 Call from SIP 398 to IP Touch 323 Check that the call is established in G729 Check audio quality Call from IP Touch 323 to SIP 398 Check that the call is established in G729 Check audio quality	☐	☒	☐	
3	Select G723 as 1st codec in TB30 Check that the call is established in G723 Check audio quality Call from IP Touch 323 to SIP 398 Check that the call is established in G723 Check audio quality	☐	☒	☐	
4	**Configure** 398 **to use VAD** **Configure IP Touch** 323 **NOT to use VAD** Call from SIP 398 to IP Touch 323 Check that the call is established in G711 A-law. Check audio quality **Configure SIP** 398 **to use VAD** **Configure IP Touch** 323 **to use VAD** Redo the same tests **Configure SIP** 398 **NOT to use VAD** **Configure IP Touch** 323 **to use VAD** Redo the same tests	☐	☒	☐	
5	**In OXO enable codec pass through for SIP phones** **Call from SIP** 397 **to SIP** 398 Check that the call is established using G.722 Check audio quality	☐	☒	☐	

6	**In OXO 1 and OXO 2 enable codec pass through for SIP phone ; direct RTP and codec pass through for SIP trunk. G722 is preferred codec in TB30** **Call from SIP** 397 **to Network SIP** 122 Check that the call is established using direct RTP in G722. Check audio quality	☐	☒	☐	
7	**In OXO 1 and OXO 2 disable direct RTP and codec pass through for SIP trunk ARS table is configured with "default" codec. G722 is preferred codec in TB30** **Call from SIP** 397 **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** **Call from SIP** 397 **to Network SIP** 122 Check that the call is established in G729. Check audio quality	☐	☒	☐	

6.3 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 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 398 call the IP Touch 322. Check that 322 is ringing. Take the call and check ring back tone audio and display.	☐	☒	☐	
2	**Call to local user with no answer** With SIP Phone 399 call the IP Touch 322. And never take the call. Check time out (if any) and display. Note that 322 don't have a Voice Mail	☐	☒	☐	322 is ringing until 399 releases the call
3	**Call to another SIP set** With the SIP phone 398 call the other SIP Phone 399 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 398 call a wrong number Check the ring back tone and display	☐	☒	☐	
5	**Call to busy user** **(SIP: "486 Busy Here")** With the SIP phone 398 call IP Touch 322, take the call and don't hang up. With other SIP phone 399 call 322 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 399 call the IP Touch 322 which is in "Out of Service State" Check the display and ring back tone	☐	☒	☐	
7	**Call to user in "Do not Disturb" (DND) state** **(SIP: "480 Temporarily not available")** Dial "*63" on the IP Touch 322 in order to enable the DND. Wait for acknowledgement ring back tone from OXO. With the SIP phone 398 call 322. Check ring back tone and display. Redial *63 on 322 to cancel the DND	☐	☒	☐	Prefix *60 to *69 is configured in the OXO for Call forwarding (see appendix C)

Test Case Id	Test Case	N/A	OK	NOK	Comment
8	**Call to local user, immediate forward (CFU). (SIP: “181 Forwarded”)(1)** On IP Touch 322 dial the *61323 (*61 + 323) to activate the CFU. Wait for acknowledgement ring back tone from OXO. With the SIP phone 398 call the 322. Check that 323 is ringing and the display. Take the call check audio and hung up. Dial *60 on 322 for forward cancellation.	☐	☒	☐	
9	**Call to local user, forward on no reply (CFNR). (1)** On IP Touch 322 configure with OMC the CFNR using dynamic routing to 323. With 398 call the 322. Check that 322 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 323 is ringing and take the call. Check the audio and display.	☐	☒	☐	OK but : When 323 answers OXO sends 181 Forwarded and 200 OK with 323 as contact but the display is not updated on SIP phone 398
10	**Call to local user, forward on busy (CFB). (1)** On IP Touch 322 dial the *62323 (*62+<target MCDU number>) to activate the CFB. Wait for acknowledgement ring back tone from OXO. With SIP phone 398 call 322 and take the call to make it busy. With other SIP phone 399 call 322. Check that 323 is ringing and take the call. Check the audio and display. Dial *60 on 322 for forward cancellation.	☐	☒	☐	OK but : When 323 answers OXO sends 181 Forwarded and 200 OK with 323 as contact but the display is not updated on SIP phone 398
11	**Call to external number (Check ring back tone, called party display)** With SIP set 398 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 398 call IP Touch 322. Take the call on 322 and never hang up, wait for timeout expire. Check that call is maintained or release.	☒	☐	☐	N/A OmniPCX Office R8.1 doesn’t implement SIP session timer for SIP phones

Notes:

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

6.4 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 feature 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 322 call SIP set 398. Check that 398 is ringing and take the call Check ring back tone and called party display. Network: with IP Touch 322 call SIP set 123 on another Node. Check that 123 is ringing and take the call. Check ring back tone and called party display.	☐	☒	☐	
2	**Local/network call to busy SIP terminal** Local: With SIP set 399 call other SIP set 398 and take the call to make it busy, don't hang up. With IP Touch 323 call 398 which is busy Check the ring back tone and display. Network: With SIP set 398 call SIP set 123 and take the call to make it busy, don't hang up. With 322 call 123 which is busy Check ring back tone and called party display.	☐	☒	☐	
3	**Local/network call to unplugged SIP terminal** Local: Unplug the 398 SIP set and call it with IP Touch 322. Check the ring back tone and display Network: Unplug the SIP set 123 and call it with 322 Check the ring back tone and display	☐	☒	☐	Call to SIP phone is not routed to its VMU until OXO detects SIP phone is unregistered (after register time out)
4A	**Local/network call to SIP terminal in Do Not Disturb (DND) mode** **By local feature if applicable:** Local: Enable DND on SIP set 398 and call it with IP Touch 322 Check the ring back tone and display Cancel the DND on 398. Network: Enable DND on SIP set 123 and call it with IP Touch 322 Check the ring back tone and display Cancel the DND on 398.	☐	☒	☐	Local: IP Touch 322 display: "released". Network: IP Touch 322 display: "unobtainable".

Test Case Id	Test Case	N/A	OK	NOK	Comment
4B	**By system feature** Local: Enable DND on SIP set 398 using the *63 prefix.. Wait for acknowledgement ring back tone from OXO. With IP Touch 322 call 398 Check the ring back tone and display Cancel the DND on 398 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 322 call 123 Check the ring back tone and display Cancel the DND on 123 using * 60 prefix.	☐	☒	☐	DND is activated and working in the call server but OXO returns a wrong SIP msg “500 Internal Server Error” : end user feedback is wrong.
5A	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to local user:** **By local feature if applicable:** Local: On SIP set 398 enable CFU to IP Touch 322 With SIP set 399 call 398. Check that 322 is ringing. Take the call and check audio and display. Disable CFU on 398. Network: On SIP set 123 enable CFU to IP Touch 102. With SIP set 398 call 123. Check that 102 is ringing. Take the call and check audio and display. Disable CFU on 123.	☐	☒	☐	Display on 399 is not updated when 322 rings and answers.
5B	**By system feature:** Local: On SIP set 398 enable CFU to IP Touch 322 using *61322 prefix (*61 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP set 399 call 398. Check that 322 is ringing. Take the call and check audio and display. Disable CFU on 398 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 399 call 123. Check that 102 is ringing. Take the call and check audio and display. Disable CFU on 123 using *60 prefix.	☐	☒	☐	CFU is activated and working in the call server but OXO returns a wrong SIP msg “500 Internal Server Error” : end user feedback is wrong. TB 30 display is not updated.
6A	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to network number:** **By local feature if applicable:** Local: On SIP Set 399 enable CFU to SIP Set122.With SIP set 398 call 399. Check that 122 is ringing. Take the call and check audio and display. Disable CFU on 399. Network: On SIP Set 398 enable CFU to IP Touch 102. With SIP Set 123 call 398. Check that 102 is ringing. Take the call and check audio and display. Disable CFU on 398.	☐	☒	☐	TB 30 display is not updated.

Test Case Id	Test Case	N/A	OK	NOK	Comment
6B	**By system feature:** <u>Local</u>: On SIP Set 398 enable CFU to SIP Set 122 using *61122 prefix (*61 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP set 399 call 398. Check that 122 is ringing. Take the call and check audio and display. Disable CFU on 398 using *60 prefix. <u>Network</u>: On SIP Set 398 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 398. Check that 102 is ringing. Take the call and check audio and display. Disable CFU on 398 using *60 prefix.	☐	☒	☐	CFU is activated and working in the call server but OXO returns a wrong SIP msg “500 Internal Server Error” : end user feedback is wrong. TB 30 display is not updated.
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 398 enable CFU to SIP set 122 With 399 call 398. Check that 122 is ringing. Take the call and check audio and display. Disable CFU on 398. <u>Network</u>: On SIP set 398 enable CFU to IP Touch 103. With SIP Set 122 call 398. Check that 103 is ringing. Take the call and check audio and display. Disable CFU on 398.	☐	☒	☐	TB 30 display is not updated.
7B	**By system feature:** <u>Local</u>: On SIP Set 399 enable CFU to SIP Set 122 using *61122 prefix (*61 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP Set 398 call 399. Check that 122 is ringing. Take the call and check audio and display. Disable CFU on 399 using *60 prefix. <u>Network</u>: On SIP Set 399 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 399. Check that 103 is ringing. Take the call and check audio and display. Disable CFU on 399 using *60 prefix	☐	☒	☐	CFU is activated and working in the call server but OXO returns a wrong SIP msg “500 Internal Server Error” : end user feedback is wrong. TB 30 display is not updated.
8A	**Local call to SIP terminal in “forward on busy” (CFB) state:** **By local feature if applicable** On SIP Set 398 enable CFB to IP Touch 322 With 398 call the voice mail at 500 to make it busy. With SIP Set 399 call 398 which is busy. Check that 322 is ringing Take the call and check audio and display. Disable CFU on 398.	☐	☒	☐	*Note: on the device CallWaiting must be set to OFF*

Test Case Id	Test Case	N/A	OK	NOK	Comment
8B	**By system feature:** On SIP Set 398 enable CFB to IP Touch 322 using *62322 prefix (*62 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With 398 call the voice mail at 500 to make it busy. With SIP Set 399 call 398 which is busy. Check that 322 is ringing Take the call and check audio and display. Disable CFB on 398 using *60 prefix.	☐	☐	☒	Call forward on busy is not working for SIP phones on OXO
9A	**Local call to SIP terminal in “forward on no reply” (CFNR)** **By local feature if applicable** On SIP Set 399 enable CFNR to IP Touch 322 With SIP Set 398 call 399. Check that 399 is ringing and don’t take the call, wait for time out (about 30 seconds). After time out expiration the 322 is ringing, take the call and check audio and display.	☐	☒	☐	TB 30 display is not updated.
9B	**By system feature**	☒	☐	☐	CNFR via prefix not available on OXO (dynamic routing has to be used)
10	**Call to busy user, Call waiting.** (Camp-on), local feature if applicable: With SIP Set 398 call other SIP Set 399 (multiline set) to make it busy, take the call and don’t hang up. With IP Touch 323 call 399 (on 399 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 399 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 323 enable mask Identity and call SIP Set 399 in order to hide 323 identity. Check that 399 is ringing, take the call and check that 323 identity is hidden.	☐	☒	☐	Display on TB 30 shows “Anonymous”
13	Display: Call to free SIP terminal from IP Touch user with a name containing non-ASCII characters (eg éëêèè). Check caller display. Check that SIP set is ringing and check on its display that the characters are correctly printed.	☐	☒	☐	
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.	☐	☐	☒	TB 30 returns 400_Bad request if the contact name/SIP display info contains specific characters like & (?

Test Case Id	Test Case	N/A	OK	NOK	Comment
15	SIP set is part of a sequential hunt group **(1)**. Call to hunt group. Check call/release. With IP Touch 322 call the sequential hunt group MCDU number 328 Check that 398 is ringing Take the call and don't hang up. And with IP Touch 323 call the sequential hunt group MCDU number 328 Check that 323 is ringing Take the call and don't hang up. And with SIP Set 397 call the sequential hunt group MCDU number 328 Check that 399 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 322 call the cyclic hunt group MCDU number 323 Check that 301 is ringing Take the call and hang up. And with 322 call the cyclic hunt group MCDU number 323 Check that 399 is ringing Take the call and hang up. And with SIP Set 397 call the cyclic hunt group MCDU number 323 Check that 398 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 323 call IP Touch 322 which is in MultiSet with SIP Set 399. Check that 399 and 322 both ringing. Take the call from 399 and check that 322 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.

6.5 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 399 call 322 take the call, check audio and display. With 399 put 322 on hold check tones and display on both and resume the call. With 322 put 399 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 399 (multi-lines) call 323 and take the call. 322 will be put on hold when making second call to 323 Put 323 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 399 switch between 322 and 323 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 322 and only 399 and 323 are in call Check that 399 & 323 are still in a call, check display.	☐	☒	☐	
2	Repeat the test 1C to 1D but using the call server feature	☒	☐	☐	Hold, enquiry, broker call functionality are not supported within call server for SIP device
3	**Three party conferences initiated from OXO set** With 322 call 398, take the call and don't release it. With 322 call 324, take the call and don't release it too. With 322 start a conference. Check that 322, 323 and 398 are in conference. Check audio and display.	☐	☒	☐	SIP device display is not updated when OXO users initiate the conference
4A	**Three party conferences initiated from SIP set with local feature** (if applicable) With 398 call 322 take the call and don't release it. With 398 call 323, take the call and don't release it too. With 398 start a conference by the local feature Check that 322, 323 and 398 are in conference.	☐	☒	☐	Display on IP Touch sets doesn't show the second user in the conference initiated on the SIP device

Test Case Id	Test Case	N/A	OK	NOK	Comment
	Check audio and display.				
4B	**Three party conferences initiated from SIP set with local feature**	☒	☐	☐	Conference feature is not supported within call server for SIP device
5	**Meet Me conference** With 399 call the Meet me Conference bridge dialing prefix 68 and follow instruction to open the bride. With 398 join the conference bridge by dialing prefix 69 and enter access code. With 322 join the conference bridge by dialing prefix 69 and enter access code. Check that 322, 398 and 399 are in conference.	☐	☒	☐	

6.6 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.

<u>**Note**: Unattended and Semi Attended transfer are not supported for SIP phones on OmniPCX Office.</u>

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	

6.7 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 398 dial “9” (attendant call prefix) Check audio and display	☐	☒	☐	Display on 398 shows internal number/name of Attendant extension
2	**2nd incoming call while in conversation with attendant** While SIP set 398 is in conversation with the attendant, from IP Touch 323 call 398 Answer the call and check audio and display	☐	☒	☐	
3	**SIP set call to attendant, attendant transfers to OXO set, semi-attended** From SIP set 398 dial “9” (attendant call prefix) and answer. Attendant transfer semi-attended to IP Touch 323 Answer the call and check audio and display	☐	☒	☐	
4	**SIP set call to attendant, attendant transfers to OXO set, attended** From SIP set 398 dial “9” (attendant call prefix) and answer Attendant transfer attended to IP Touch 323 Check audio and display	☐	☒	☐	
5	**OXO set calls to attendant, attendant transfers to SIP set, attended** From IP Touch 323 dial “9” (attendant call prefix) and answer Attendant transfer attended to SIP set 398 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 398 and transfer attended Check audio and display	☐	☒	☐	
7	**SIP set call to attendant, attendant transfers to External** From SIP set 398, 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.	☐	☒	☐	

6.8 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 398, 399 and OXO 322.

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 399 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 398 call the Voice Mail at 500. Follow the instructions in order to send a voice message in SIP set 399 boxes. Check that the MWI on 399 is activated.	☐	☒	☐	There is no count of messages on the display. Notification is only for messages.
3	Message consultation With SIP set 399 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 399 after message cancellation.	☐	☒	☐	
4	SIP call to a OXO user forwarded to Voice Mail Forward the IP Touch 322 to Voice Mail by dialing *61500 (*61 prefix + <Voice Mail number>). With SIP set 399 call 322 and check that you are immediately forwarded to Voice Mail. Check that you can leave a message On 322 disable Voice Mail forwarding with *60 prefix.	☐	☒	☐	
5	OXO set call to a SIP user forwarded to Voice Mail Forward the SIP set 399 to Voice Mail by dialing *61500 (*61 prefix + <Voice Mail number>). With IP Touch 322 call 399 and check that you are immediately forwarded to Voice Mail. Check that you can leave a message On 399 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.

6.9 Defence

Checks how 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.	☐	☒	☐	ISDN call is rerouted to attendant after register time-out

7 Appendix A: Partner Application description

TB30:

Configure phone using web interface

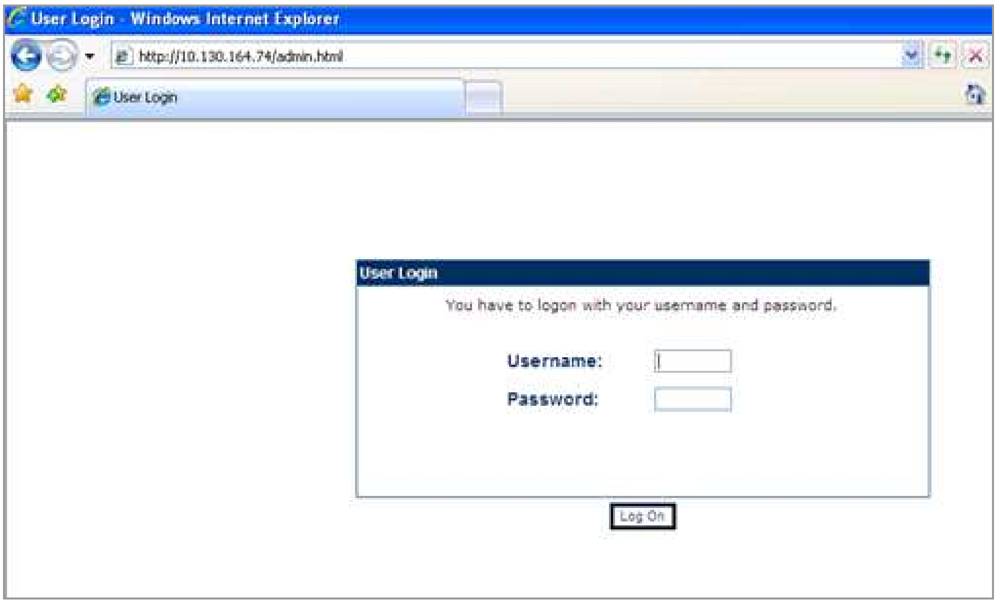
Find the phone IP address

1. Press menu key
2. Choose User with soft keys
3. Use the Navigator keys to scroll down the menu to Information
4. Press Select soft key
5. The IP address is displayed

8 Appendix B: Partner Application: configuration requirements

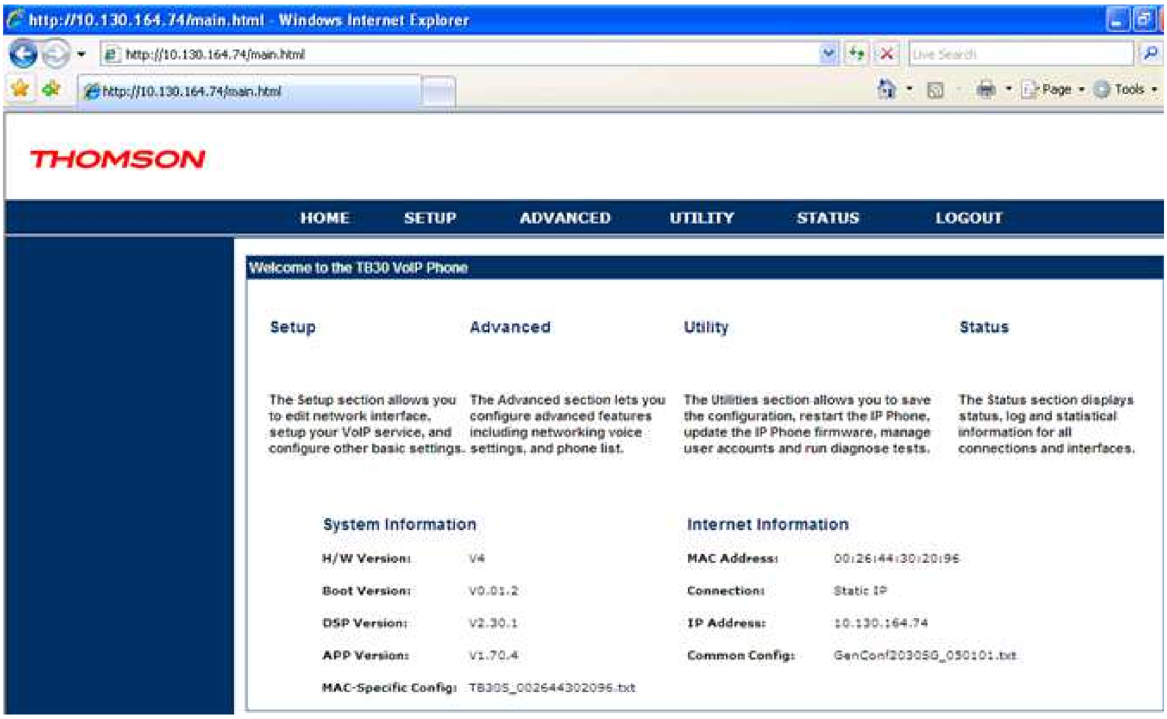
Access to Admin Homepage (web interface)

1. Open a web browser (Firefox, Internet Explorer, Safari…)
2. Enter the TB30 IP address in the address bar with /admin.html appended at the end of it. Ex: http://10.130.164.74/admin.html



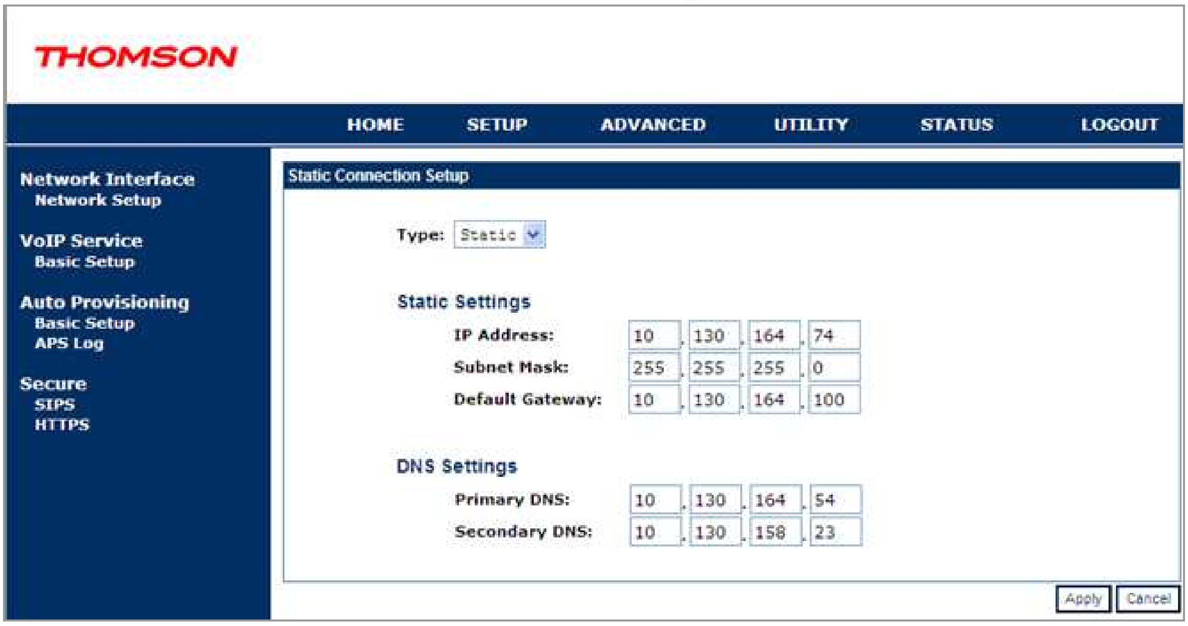
3. Enter the administrator login “administrator”
4. Enter the administrator password (default value is 784518)
5. You will access the Homepage of the TB30

Main Page Details

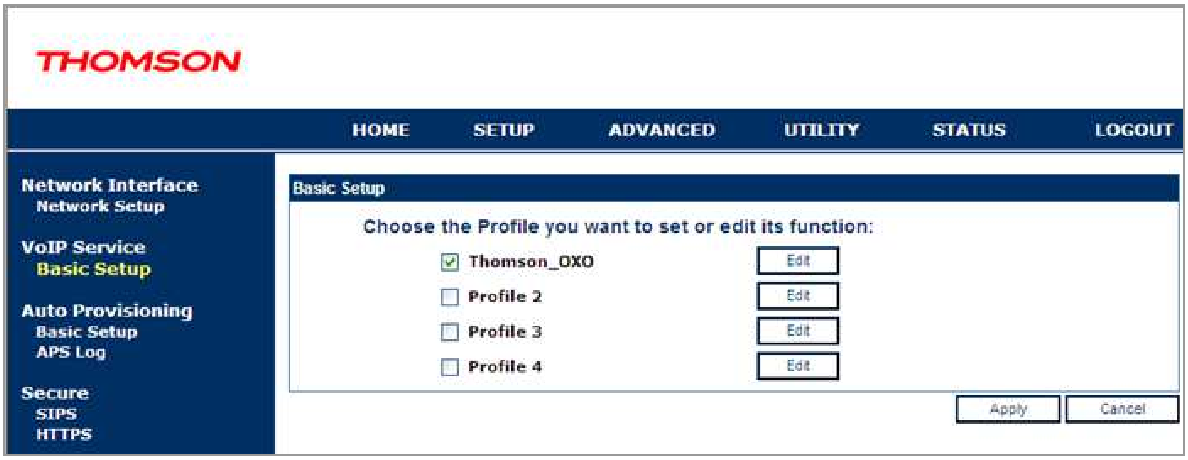


Network Configuration Details

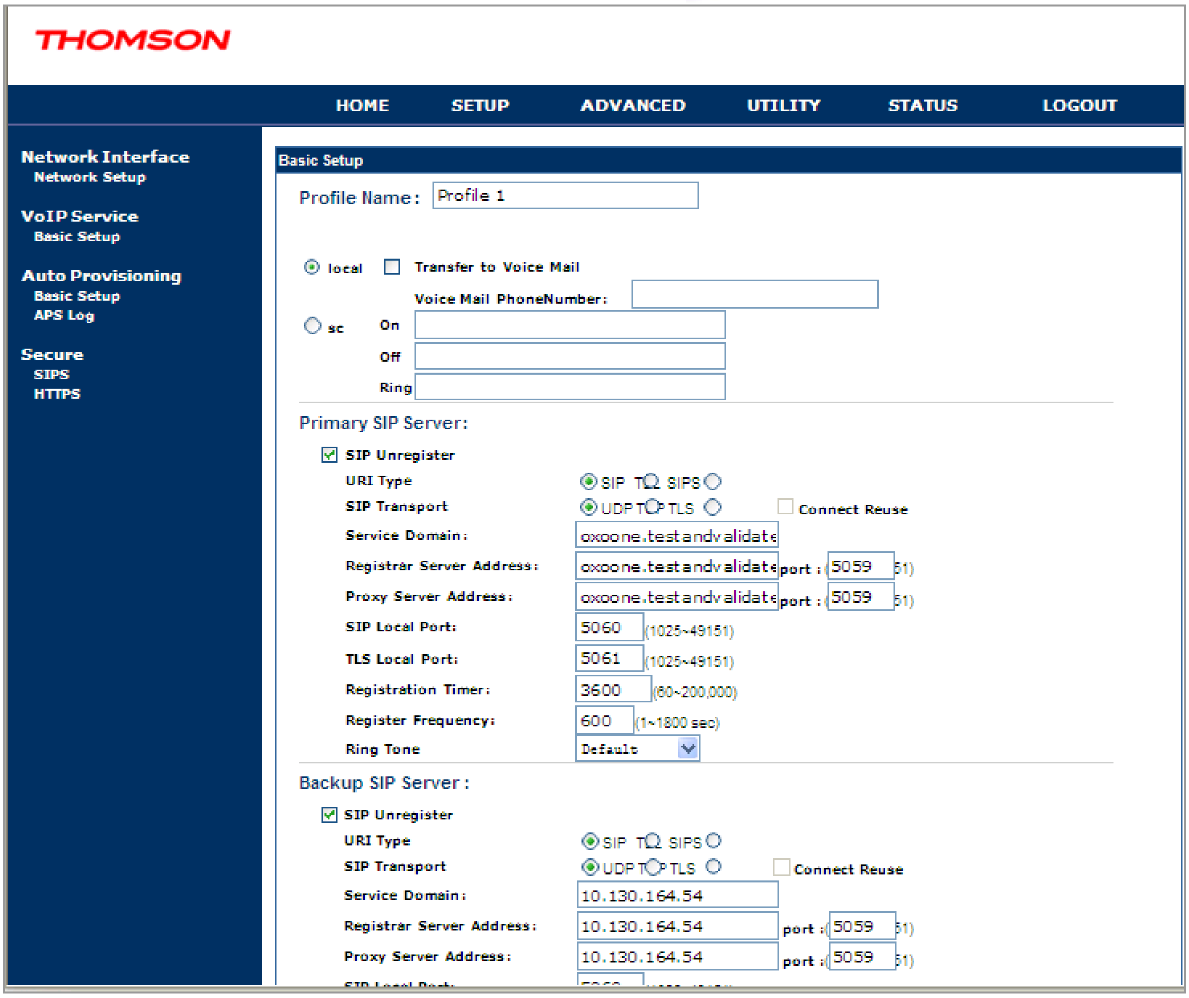
Configuring Various Network parameters



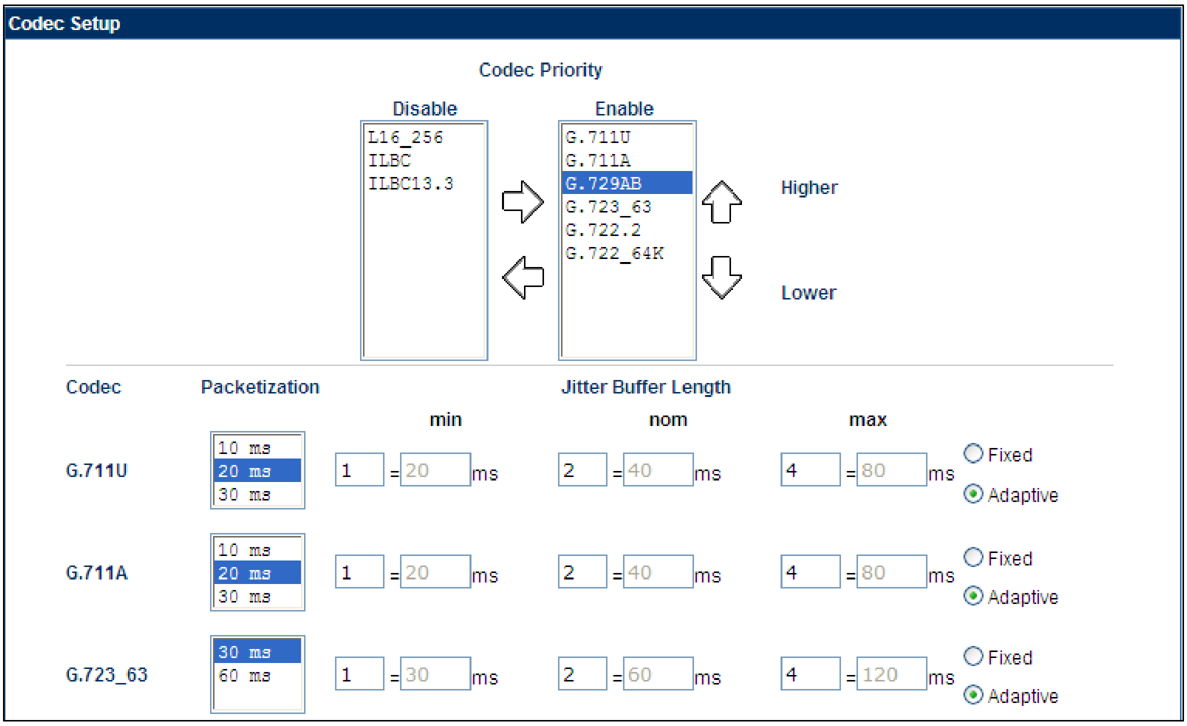
SIP parameters configuration



Enable SIP profile and configure various parameters like registrar server, Proxy server, SIP transport protocol, Phone number, authentication Id and password.
The proxy and registrar server information can be name based or IP address based.



CODEC configuration

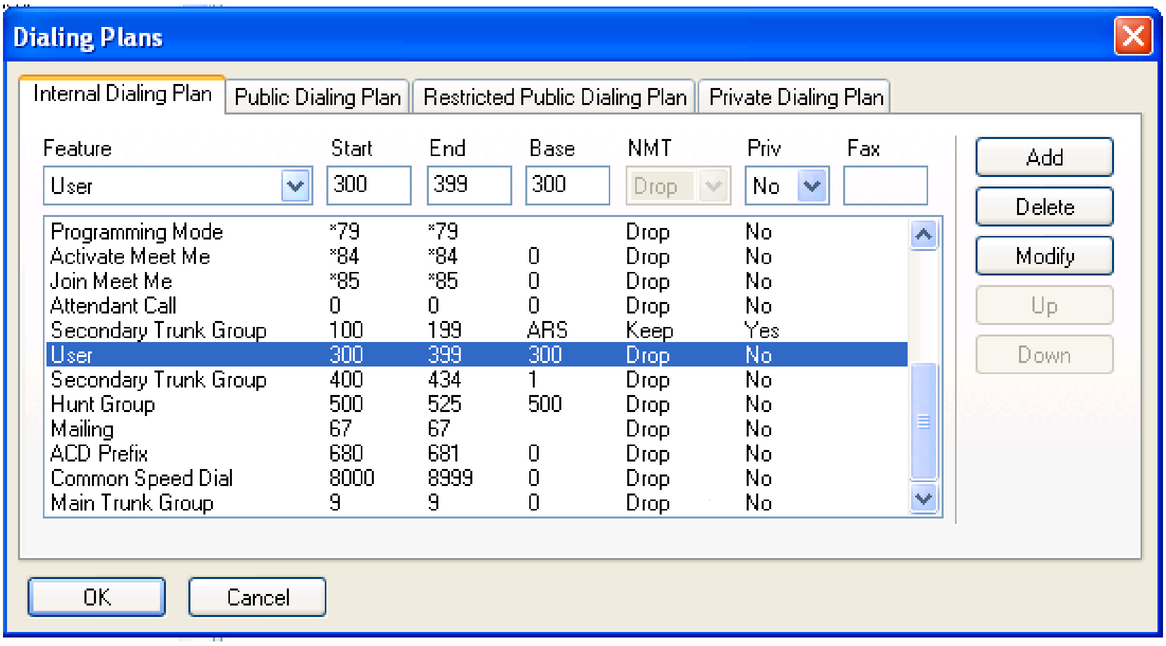


Enable the required CODECs enabled and keep the priority of the CODEC as required.

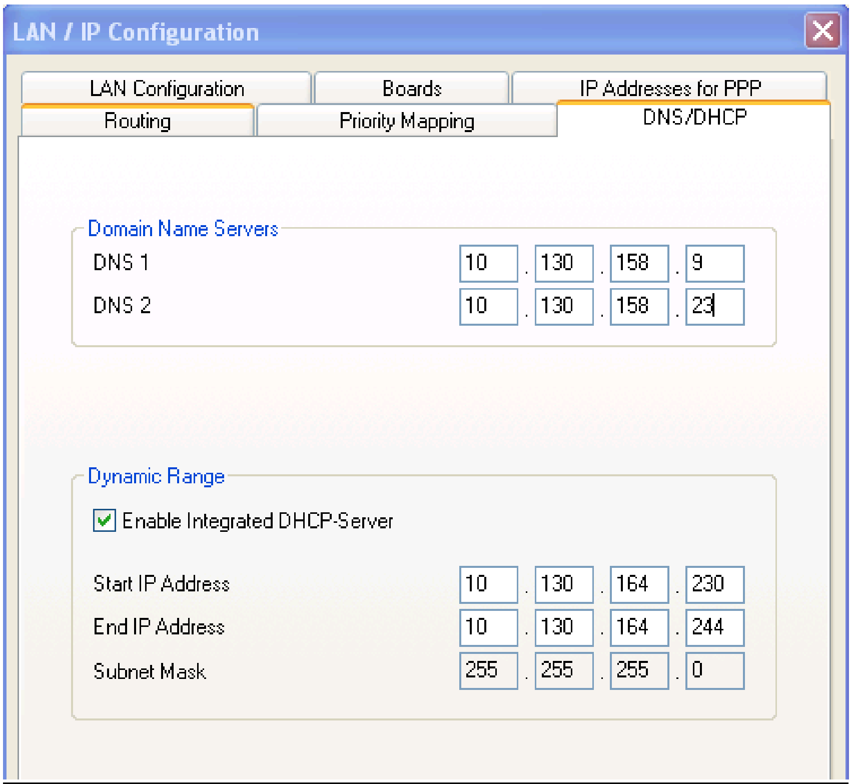
9 Appendix C: Alcatel-Lucent Communication Platform: configuration requirements

OXO Configuration

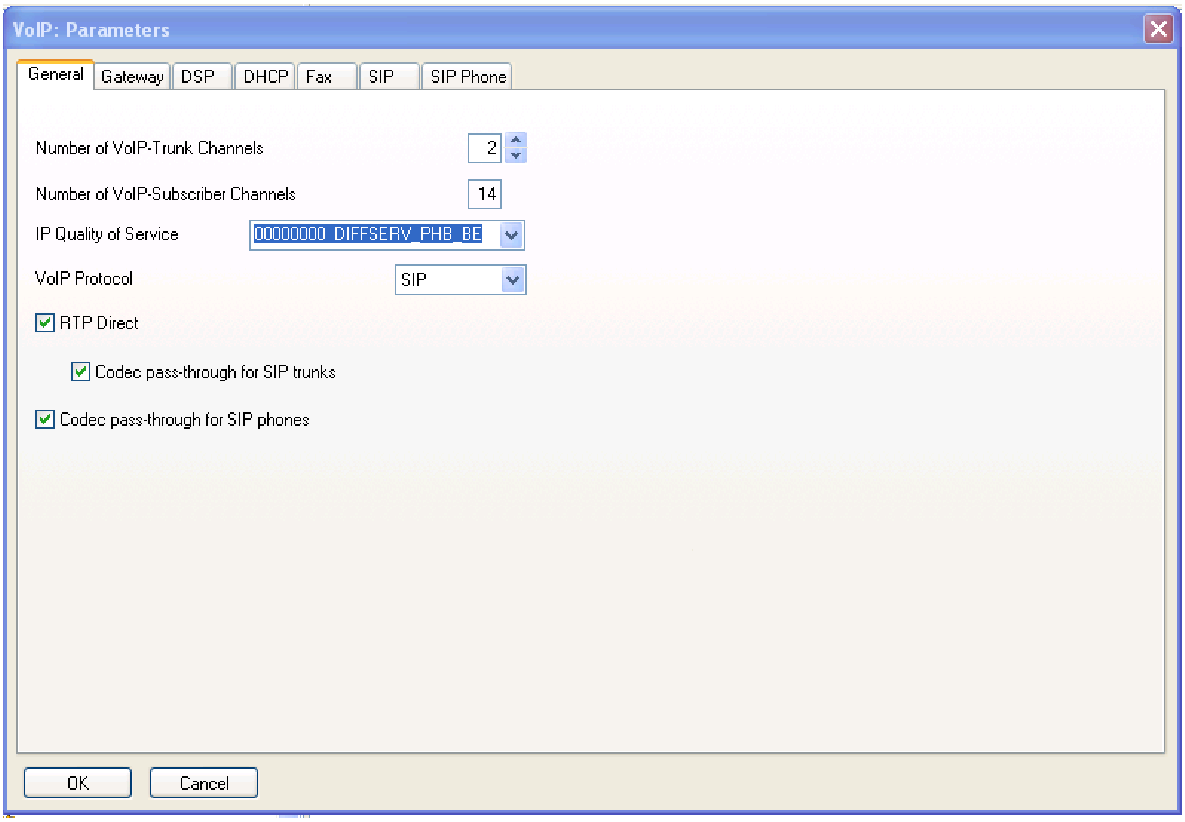
1. Dialing Plan:

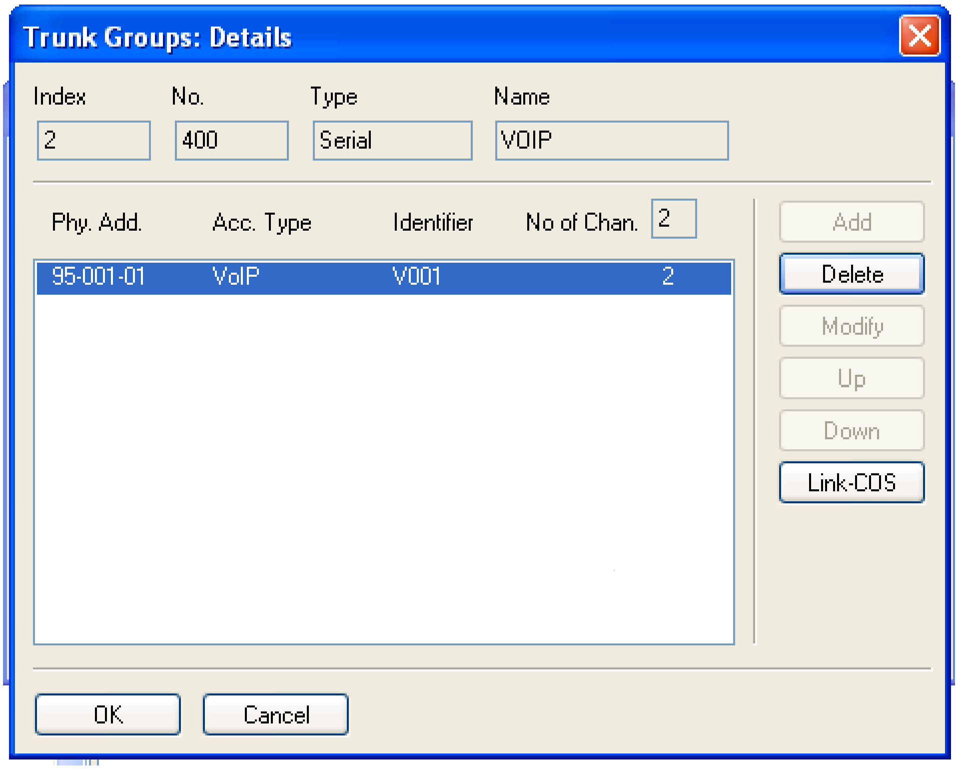


2. DNS/DHCP Configuration:

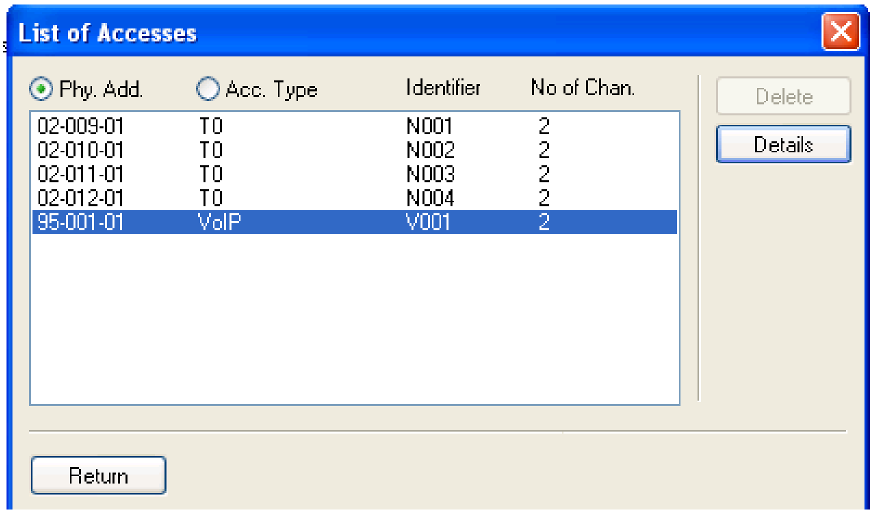


3. Trunk Configuration:

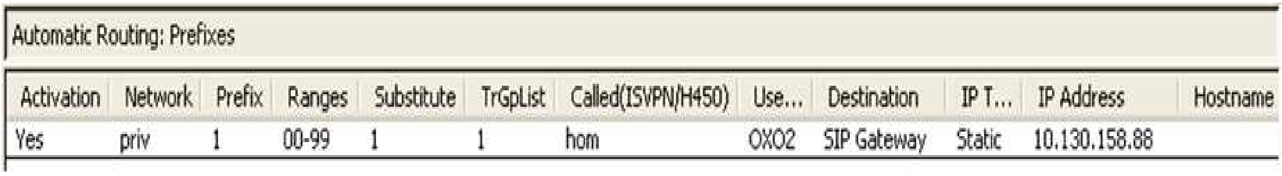




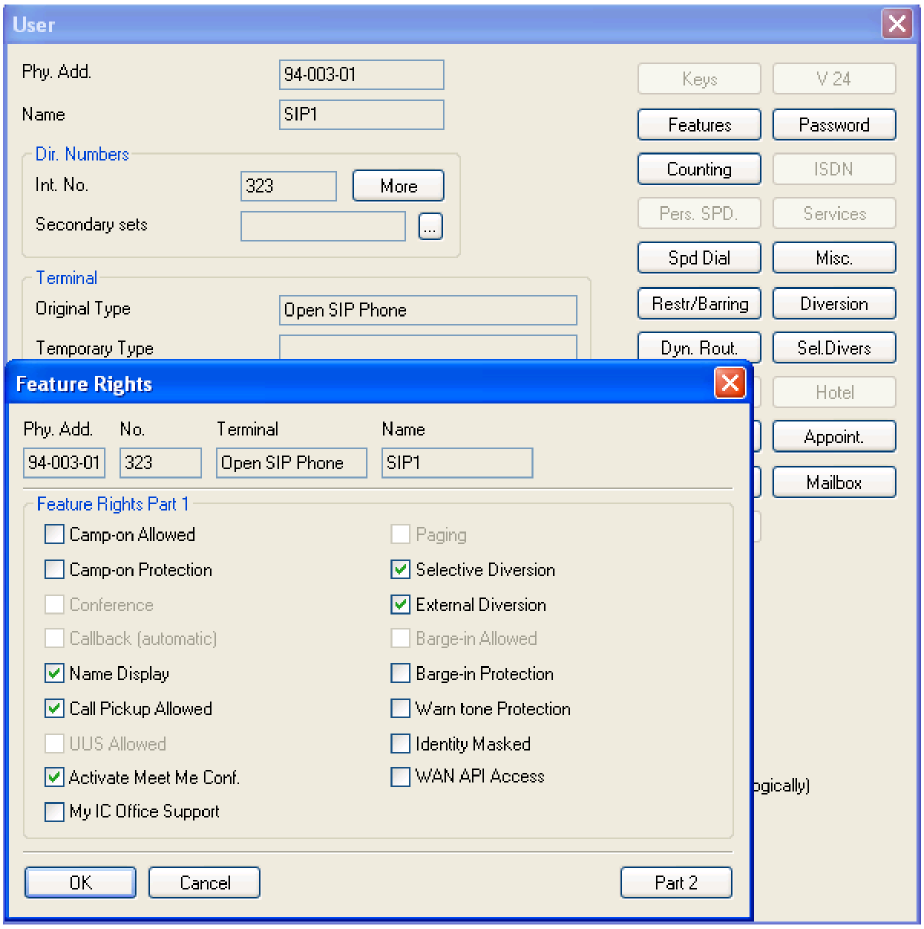
4. Trunk Access:



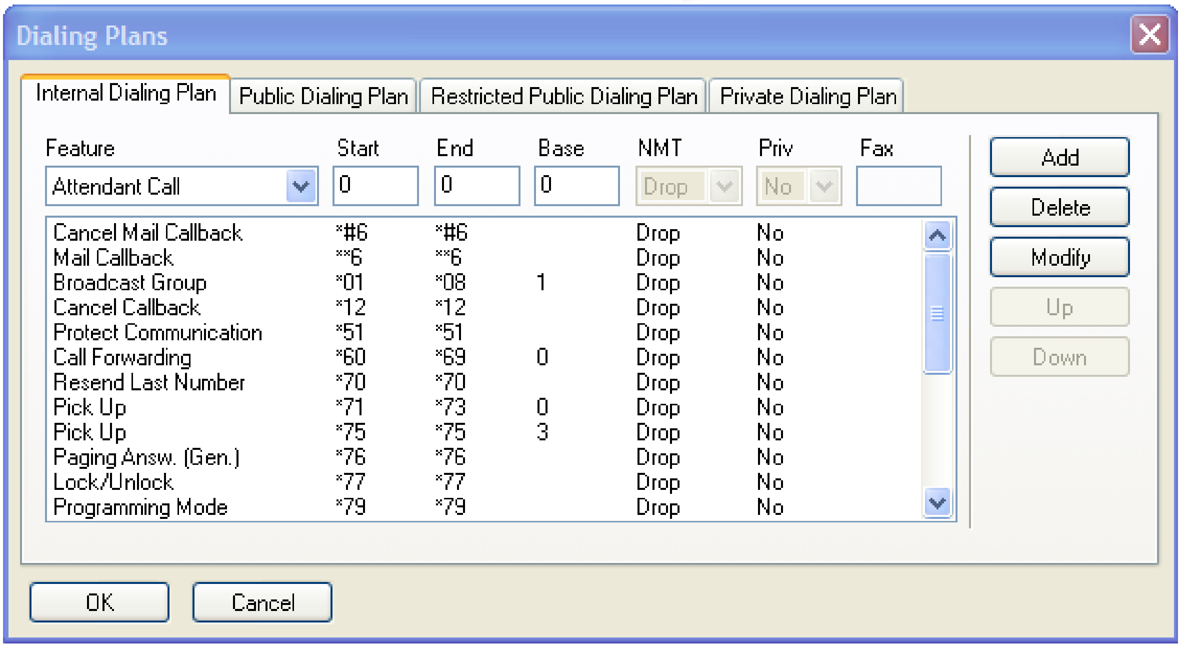
5. Network Call Configuration:

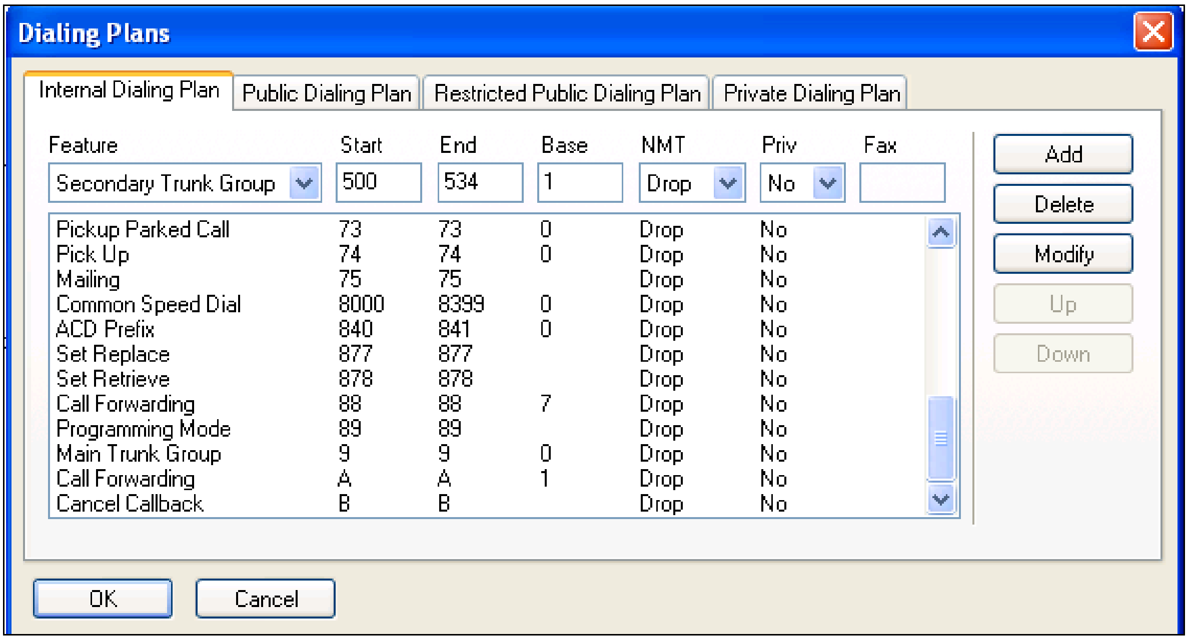


6. SIP Set Configuration:



7. Lists of OXO prefixes used in tests


Dialing Plans
Internal Dialing Plan
Public Dialing Plan
Restricted Public Dialing Plan
Private Dialing Plan
Feature Start End Base NMT Priv Fax
Attendant Call 0 0 0 Drop No
Cancel Mail Callback *#6 *#6 Drop No
Mail Callback **6 **6 Drop No
Broadcast Group *01 *08 1 Drop No
Cancel Callback *12 *12 Drop No
Protect Communication *51 *51 Drop No
Call Forwarding *60 *69 0 Drop No
Resend Last Number *70 *70 Drop No
Pick Up *71 *73 0 Drop No
Pick Up *75 *75 3 Drop No
Paging Answ. (Gen.) *76 *76 Drop No
Lock/Unlock *77 *77 Drop No
Programming Mode *79 *79 Drop No
Add
Delete
Modify
Up
Down
OK
Cancel


Dialing Plans
Internal Dialing Plan
Public Dialing Plan
Restricted Public Dialing Plan
Private Dialing Plan
Feature Start End Base NMT Priv Fax
Secondary Trunk Group 500 534 1 Drop No
Pickup Parked Call 73 73 0 Drop No
Pick Up 74 74 0 Drop No
Mailing 75 75 Drop No
Common Speed Dial 8000 8399 0 Drop No
ACD Prefix 840 841 0 Drop No
Set Replace 877 877 Drop No
Set Retrieve 878 878 Drop No
Call Forwarding 88 88 7 Drop No
Programming Mode 89 89 Drop No
Main Trunk Group 9 9 0 Drop No
Call Forwarding A A 1 Drop No
Cancel Callback B B Drop No
Add
Delete
Modify
Up
Down
OK
Cancel

10 Appendix D: Partner escalation process

In Case of problem please contact

Cyril Cousteur:
Email: Cyril.Cousteur@thomson.net
Tel: +852 2589 9358 (Hong Kong)

For more update information on phone & contact:
http://www.thomsonbroadbandpartner.com/telephony-solutions/products/voice-over-iptelephones.php

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 Allcatel-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

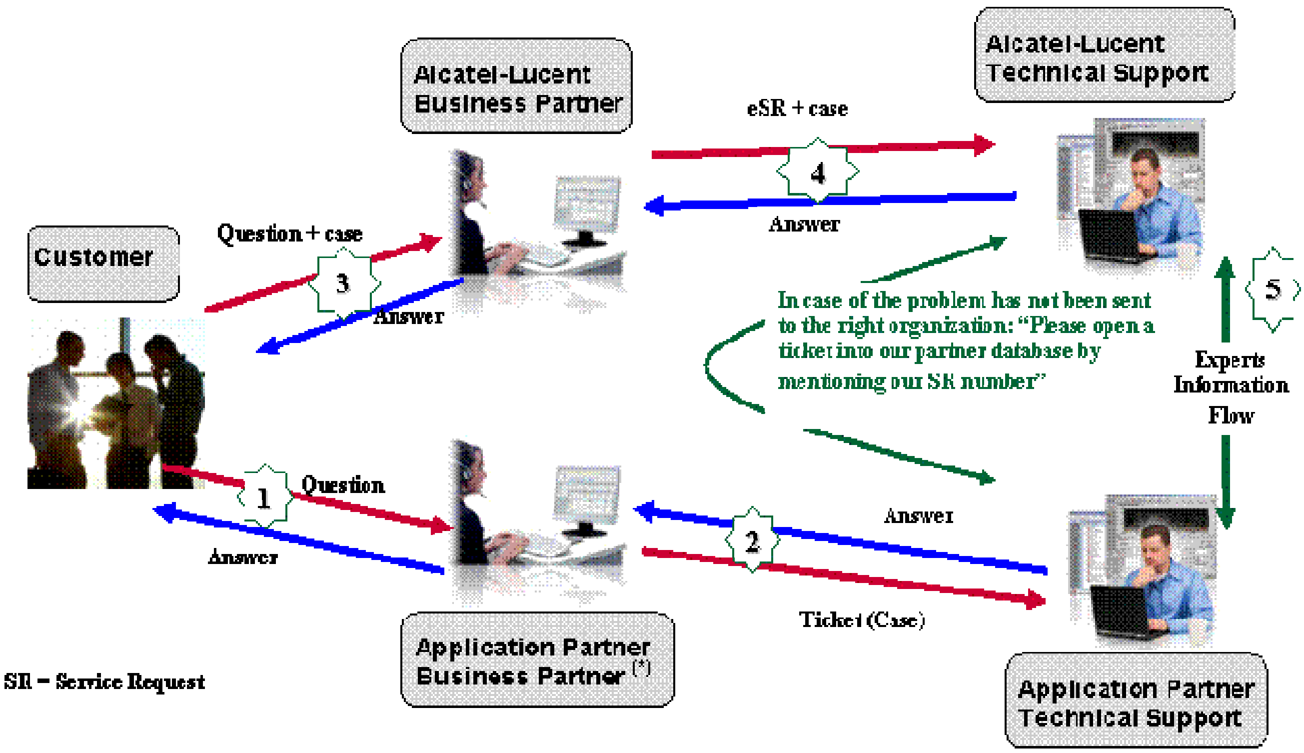
You can access the Alcatel-Lucent website at this URL: http://www.Alcatel-Lucent.com/

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:

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