Alliance & Application Partner Program
Inter-Working Report

Partner: AUDIOCODES
Application type: Media gateway
Application name: MediaPack MP11x (with SAS)

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	February, 2010

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Partner's representative	Christophe Hubaut

Alcatel-Lucent Communication Platform	OmniPCX Enterprise
Alcatel-Lucent compatibility release	R9.0 (H1.301.35)
Partner's application version	5.60A.025.005

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Historic

Edition 2: add of emergency calls configuration and tests – *03-02-2010*
Edition 1: creation of the document – *25-08-2009*

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 Alliance & 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.

2 Application information

Application type: Telephone Adapter / VoIP Gateway for Analog equipments

Application commercial name: MediaPack 118

Application version: 5.60A.025.005

Interface type: SIP

Brief application description:

AudioCodes MP118 is a telephone adapter that allows connecting ordinary analog telephones or fax machines to a Voice over Broadband service. It is typically adapted for Branch Offices.
The MP118 connects to a Service Provider by using its IP uplink connection. It proposes up to 8 VoIP ports for connecting up to 4 analog sets or faxes and 4 PSTN lines.
MP118 also supports the SIP protocol, used in the present case for communicating with OXE. The equipments connected on the MP118 ports will therefore be declared as SIP terminals (SIP extension for analog phones and SIP device for fax) and will register on OXE.
MP118 also includes the SAS (Stand Alone Survivability) feature based on SIP B2BUA (Back to Back User Agent) functionality, targeting customers of Enterprise branch offices. SAS enables the service backup for SIP clients such as SIP IP Phones and SIP Soft Phones in the case of a network failure. This backup is performed by the Media Gateway installed as a CPE in the customer premises or branch office.
In our case, when the IP link is broken between the headquarters OXE and the branch office, the branch office IPTouch switch to SIP and register to the MP118 allowing them to be able to make and receive calls (inside the branch office and also with the external world thanks to the MP118 PSTN connection).

Only the MP118 hardware is tested in this document but the behavior should be the same with all the MP11x family.

3 Tests environment

3.1 Architecture

Figure 1 Tests environment

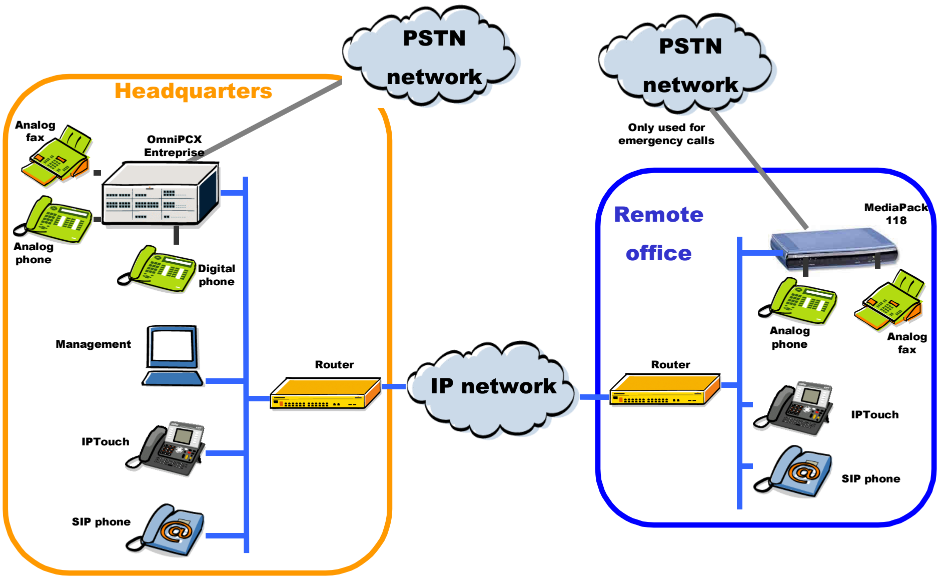
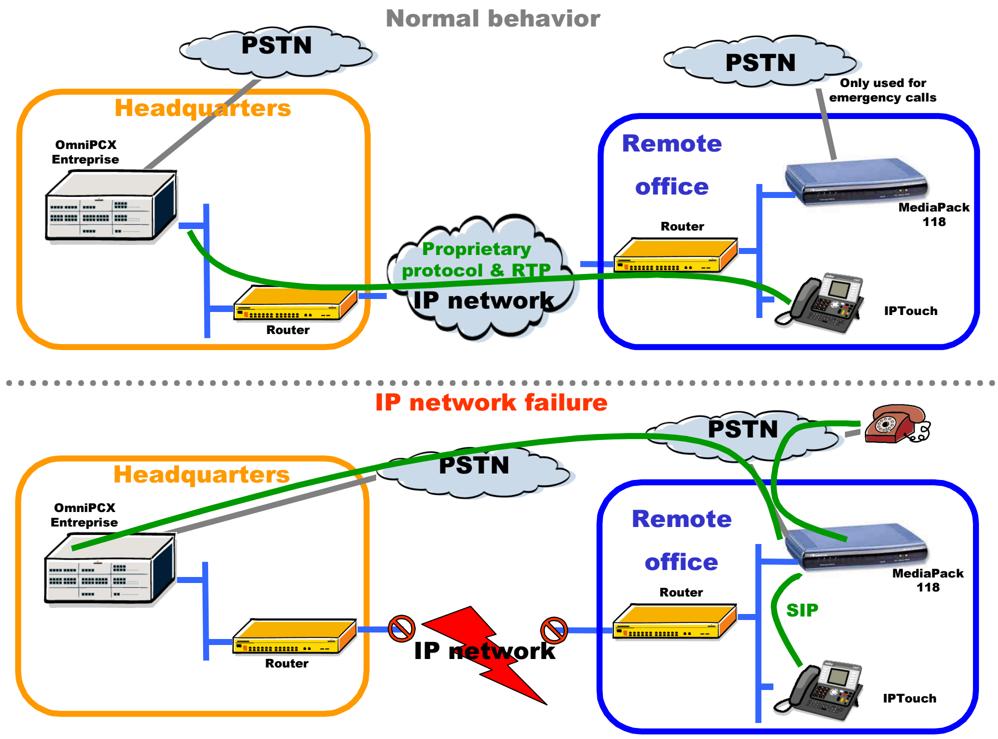


Figure 2 IPTouch SIP survivability



There are two modes:

- Normal behavior

  The MP118 is used as an analog / SIP gateway for the remote office analog phones and fax machines. These equipments are seen as SIP end points by the OXE Call Server.
  All the calls use the IP network and are handled by the OXE. The PSTN network is not used by the MP118 except for emergency outgoing calls (emergency calls are not tested in this edition).

- In case of IP network failure,

  The MP118 is used as a SIP proxy. The IPTouch have lost the connection to the OXE Call Servers and register to the MP118 in SIP mode.
  Calls are handled by the MP118.
  In this case, calls between the remote office and the headquarters use the PSTN network.
  Remote office users (analog phones and fax machines, IPTouch) can make (receive) calls to (from) other remote office users and also to (from) external PSTN users thanks to the MP118 FXO port.

3.2 Network configuration

Headquarters:

- OXE OmniPCX Enterprise (common hardware) with two CPU in spatial redundancy
  - IP address: 192.168.24.6 (first CPU role address) and 192.168.25.6 (second CPU role address)
  - SIP domaine name: Etoilenoire
  - Attendant: directory number 70000
  - 4645 Voice mail: directory number 20000, hosted on first OXE CPU
  - IPTouch phone: directory number 21017
  - UA phone: directory number 21019
  - PSTN (ISDN T2 access): number +33-388668490 (to attendant) and +33-388668491 (to IPTouch 21017). Prefix: 0
  - Analog fax: directory number 26002
  - IP domain 1: IP address range 192.168.24.105 to 192.168.24.109. This domain is rescueded by the MP118
  - Emergency number: 911

Remote office:

- MP18
  - IP address: 192.168.24.105
  - Analog phones: directory numbers 22018 (FXS port 1), 22019 (FXS port 2), 22020 (FXS port 3)
  - IPTouch phones: directory numbers 21013 (IP address 192.168.24.106) and 21028 (IP address 192.168.24.107)
  - Fax: directory number 26005 (FXS port 4)
  - PSTN (FXO port 5): number +33-388673291 (to analog phone 22018 or IPTouch phone 21013 depending on the test). Prefix: 0
  - Emergency number: 911

3.3 Hardware configuration

- Alcatel OmniPCX Enterprise :
    - Common hardware : CallServer, MediaGateway, MIX board (isdn, digital, analog), digital and analog sets, 4008, 4018, 4038 and 4068 IPTouch Extended Edition

- Application platform: MP118 (4 FXS and 4 FXO ports) tests are detailled in this document but all the MP11x family should have the same behavior.

3.4 Software configuration

- Alcatel Communication Platform: OmniPCX Enterprise (H1.301.35)
- Application platform: MP118 5.60A.025.005

4 Summary of test results

4.1 Summary of main functions supported

Features	Status	Comments
6.1 IP link between headquarters and remote office loss / recover	OK	
6.2 MP118 used as analog / SIP gateway		
6.2.1 Analog equipment initialization, SIP registration and authentication	OK	
6.2.2 Audio codec negociation	OK	
6.2.3 Defense (call server switch over and reboot)	OK	
6.2.4 Basic calls	OK	
6.2.5 Telephonic features		
Forward	OK but	Forward on busy is not working for an analog phone
Hold and broker call	OK	
Transfer	OK	Only transfer during ringing is possible
Conference	NA	Feature not available on the analog phone
Voice mail	OK but	Message deposit and interaction with voice mail is workind. New message signalization by led is not working on the analog phone (ok for the tone signalization)
Do not disturb	OK	
Wake up	OK	
Call back	OK but	Leaving a call back is working. But the call not established when the analog phone is called by a call back.
6.2.6 Fax	OK	
6.2.7 Emergency calls	OK	
6.3 MP118 used as SIP proxy (in case of IP network failure)		

Features	Status	Comments
6.3.1 SIP registration and authentication	OK	IPTouch SIP authentication is not supported by the Audiocodes equipment
6.3.2 Audio codec negociation	OK	
6.3.3 Defense (Audiocodes equipement and IPTouch reboot)	OK	
6.3.4 Basic calls	OK	
6.3.5 Telephonic features		
Forward	OK but	OK for the IPTouch Not OK for the analog phone: not supported by Audiocodes Only immediate forward is possible for the IPTouch
Hold and broker call	OK	There is no on hold feedback for the headqaurters or PSTN user when put on hold by an analog phone located in the remote office
Transfer	OK but	For blind transfer, IPTouch key has to be used (hook on does not work). Blind transfer is not possible when calling an analog phone while already in conversation with another analog phone. Consultative transfer is OK
Conference	NA	Feature not available on the IPTouch nor on the analog phone
Voice mail	OK	
Do not disturb	OK	
6.3.7 Emergency calls	OK	

4.2 Summary of problems

None

4.3 Summary of limitations

MP118 used as analog / SIP gateway (link between headquarters and remote office is up and running)

- In spatial redundancy, when switching from OXE main to standby CPU, next outgoing calls from the remote office analog phone fails. Incoming calls are OK. After SIP re-registration everything is OK. Thus registration timers have to be adjusted in consequence,
- The call to an analog phone forwarded on busy or on busy / no answer is not forwarded by the OXE,
- Consultative transfer feature is not available on the analog phone,
- Conference feature is not available on the analog phone,
- Led voice mail new message signalization on the analog phone is not working. Tone generation when picking up is working,
- The call back to an analog phone fails (analog phone picks up but the conversation is not established).

MP118 used as SIP proxy (IP network failure between headquarters and remote office)

- IPTouch SIP authentication feature is not available in the Audiocodes equipment,
- Call to a forwarded analog phone is not processed by the Audiocodes equipment,
- No on hold tone an analog phone in the remote office, phone located in the headquartes and PSTN user when put on hold by a remote office phone.
- When an IPTouch calls another analog phone located behind the MP118 while already in conversation with another analog phone located behind the MP118, there is no possibility to do a blind transfer. Call has to be picked up by the analog phone to allow the IPTouch to do the transfer.
- Conference feature not available on the analog phone.

4.4 Notes, remarks

- Only the MP118 hardware has been tested. Behavior should be the same with all the MP11x family,
- When using OXE prefix to set a forward, Do not disturb, wake-up, there is no indication on the remote office analog sets. Only tones are heard during programmation. There is also no indication when an OXE forward is set,
- Even if some telephonic features may be activated using Audiocodes phone local possibilities (forward, do not disturb), it is mandatory to use the OmniPCX Entreprise features (prefix and suffix).
- When in analog / SIP gateway mode (network link between headquarters and remote office up and running), it is possible to use the Audiocodes MP118 FXO connection to the public network as a proximity outgoing public network trunk for the remote office users (IPTouch, analog phones). In this case, remote office phone calls using the public network prefix are routed by the OXE to the SIP trunk to the Audiocodes MP118 FXO trunk and not to the headquarters public network trunk. The OXE configuration is exactly the same as for the emergency calls except that the ARS prefix has to be changed from 911 to 9 (if 9 is the public network access prefix).

5 Test Scenarios

5.1 Test procedure

Step	Action	Result	Comment

Step: a test may comprise multiple steps depending on its complexity. Each step has to be completed successfully in order to conform to the test. Step 0 when present represents the initial state for all the following steps.
Action: describes which action to realize in order to set-up the conditions of the test.
Result: describes the result of the test from an external point of view. If it is positive, it describes which application's trigger was checked. If it is negative, it describes as precisely as possible the problem. If the step within this test is not applicable to this application: **NA**. This has to be filled in only if the test is checked as mandatory in the applicability box. In that case, the column comment must indicate the reason of the non-applicability (e.g.: service not supported). **NT** means not tested.
Comment: this column has to be filled in when a problem occurs during the test. It must contain a high level evaluation of the localization of the responsibility: Alcatel-Lucent or the Partner.

☞ **it is not intended during this test session to debug and fix problems**.

5.2 Result template

The results table must be formatted as indicated in the example below:

Step	Action	Result	Comment
1	**.** action 1	OK	
2	**.** action 2	OK	The application waits for PBX timer or phone set hangs up
3	**.** action 3	OK	
4	**.** action 4		Relevant only if the CTI interface is a direct CSTA link
5	**.** action 5	NOK	No indication, no error message
...	...		

6 Testing

6.1 IP link between headquarters and remote office loss / recover

These tests check the switch between the two modes (IP link between headquarters and remote office active and inactive).

6.1.1 Switch

6.1.1.1 Test objectives

Phone behaviors are checked when the headquarters to remote office IP link goes down and up.

6.1.1.2 Test procedure

Step	Action	Result	Comment
1	IP link is down Check the remote office IPTouch restart and register (in SIP mode) to the MP118. Check they can receive and make calls from/to ▪ a PSTN user, ▪ an analog phone (located behind the MP118), ▪ another IPTouch located in the remote office. Check the analog phone located behind the MP118 can receive and make from/to a PSTN user.	OK	

Step	Action	Result	Comment
2	IP link is up Check the remote office IPTouch restart and register (in proprietary mode) to the headquarters OXE. Check they can receive and make calls from/to ▪ a headquarters phone, ▪ an analog phone (located behind the MP118).	OK	

6.2 MP118 used as analog / SIP gateway

This is the normal mode. The IP network is up and used for the calls between the remote office and the headquarters. The MP118 acts as an analog / SIP gateway.

6.2.1 Analog equipment initialization, SIP registration and authentication

6.2.1.1 Test objectives

These tests check that the equipments (analog phones and fax machines) are able to register to the OXE with and without SIP authentication. The MP118 SIP configuration possibilities are also tested especialy for the OXE Call Server redundancy support (alternate proxys, DNS).

6.2.1.2 Test procedure

Step	Action	Result	Comment
1	Analog phone set registration to OXE using alternate proxys The MP118 is configured to use proxy servers address. The main and alternate proxy address are the OXE CPU address. Tests are performed when first Call Server is activ and then when second Call Server is activ.	OK	This method is described in this document (see configuration annex).
2	Analog phone set registration to OXE using DNS The MP118 is configured to use a domain name as registrar / proxy server address. The DNS IP address are the OXE CPU address. Tests are performed when first Call Server is activ and then when second Call Server is activ.	OK	
3	Analog phone set registration to OXE using SIP digest authentication SIP digest authentication is activated on OXE and phone side (MP118). Check also that outgoing call is authenticated.	OK	

6.2.2 Audio codec negociation

6.2.2.1 Test objectives

These tests check that the MP118 is using the configured audio parameters (codec, framing, VAD).
The MP118 and OXE negotiate the appropriate codec during a basic call between a UA phone and an analog phone behind the MP118. Same test also between an IPPhone and the analog phone.

6.2.2.2 Test procedure

Step	Action	Result	Comment
1	The MP118 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 Alaw	OK	
2	The MP118 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	OK	
3	The MP118 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	OK	
4	The MP118 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 Alaw	OK	
5	The MP118 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.	OK	
6	The MP118 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.	OK	
7	Repeat previous 9 tests by changing Alaw to µ law	OK	Only step 1 and 4 have been performed.
8	Codec selection. The MP118 and OXE do not have any common codec. For example, the MP118 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.	OK	OXE answers 415 Unsupported media type MP118 answers 488 Not acceptable here

6.2.3 Defense

6.2.3.1 Test objectives

These tests check the MP118 defenses against perturbations and OXE Call Servers switch over.

6.2.3.2 Test procedure

Step	Action	Result	Comment
1	OXE Call Server CPU switch over while the analog phones behind the MP118 are in idle. Check the behavior after a switch from the OXE main to standby CPU. The analog phone must be able to make and receive a call after the switch over.	OK	
2	OXE Call Server CPU switch over while the analog phones behind the MP118 are in conversation with an IPTouch. Check the behavior after a switch from the OXE main to standby CPU. The call is still activ. The phone can make and receive a second call and switch from one to another. After on hook, the analog phone must be able to make and receive a call after the switch over.	OK but	Conversation remains activ. When the analog phone hooks on, the MP118 sends the BYE to the wrong CPU. But we can imagine that the remote side will hook o by itself also.
3	OXE Call Server reboot while the analog phones behind the MP118 are in idle. 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 analog phone is able to make and receive a call.	OK	
4	OXE Call Server reboot while the analog phones behind the MP118 are in conversation with an IPTouch. Check the behavior when the OXE Call Server reboots (without standby CPU). The call is released. As soon as the Call Server is running again, the analog phone is able to make and receive a call. Check also the behavior when in communication with another analog set behind the MP118 (call remains established).	OK	

6.2.4 Basic calls

6.2.4.1 Test objectives

These tests check the analog phone located behind the MP118 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.

The description of the following tests are detailed based on the point of view of a user using an analog phone located behind the MP118.
For example, an outgoing call to an IPTouch means a call made by this analog phone to the an IPTouch located in the remote office or headquarters.

6.2.4.2 Test procedure

Step	Action	Result	Comment
1	Call from and to a SIP phone. The analog phone calls a SIP phone. The analog phone is called by a SIP phone. In both cases, check the display and audio during all steps (dialing, ring back tone, conversation, release).	OK	
2	Call from and to an IPTouch. Repeat **1** with an IPTouch.	OK	
3	Call from and to an UA phone. Repeat **1** with an UA phone.	OK	
4	Call from and to an analog phone. Repeat **1** with an analog phone located in the headquarters. Repeat **1** with an analog phone located in the remote office behind the MP118.	OK	
5	Incoming call released by the caller during ringing. The caller releases the incoming call to the analog phone before the callee takes the call.	OK	
6	Outgoing call released by the caller during ringing. The caller releases the outgoing call from the analog phone before the callee takes the call.	OK	

Step	Action	Result	Comment
7	Incoming call rejected by the callee during ringing. The callee rejects the incoming call to the analog phone during ringing.	NA	Feature not available on the analog phone.
8	Outgoing call rejected by the callee during ringing. The callee rejects the outgoing call from the analog phone during ringing.	NA	Feature not available on the IPTouch.
9	Call released by the analog phone. The analog phone releases the call after a conversation period.	OK	
10	Call released by the other phone. The other phone releases the call after a conversation period.	OK	
11	Incoming call presentation while already in conversation. The analog phone is already in conversation and receives a new incoming call. Check the display (new call presentation) and audio (new call signalization).	OK	There is a beep and tone played one time by the analog phone.
12	Call from and to an external number (T0/T2) Call is properly established.	OK	
13	Call from and to an attendant Call is properly established.	OK	
14	Incoming external call (T0/T2 for example) to an attendant phone set which tranfers the call to the analog phone. Transfer is done while the analog 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.	OK	OK for transfer after pick up. Ok for blind transfer/on ringing.
15	Outgoing call from a analog phone to an attendant with transfers to an external call (T0/T2 for example. Call is properly established.	OK	OK for transfer after pick up. Ok for blind transfer/on ringing.
16	Dialing break The analog 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.	OK	

6.2.5 Telephonic features

6.2.5.1 Test objectives

These tests check the analog phone (located behind the MP118) behavior during OXE telephonic feature use like forward, on hold, transfer, voice mail interactions, conference.

Tests ae also done using the MP118 local feature possibilities.
For example, setting an immediate forward for calls to the analog phone.

6.2.5.2 Test procedure

Step	Action	Result	Comment
1	Immediate forward to another phone. Prefix: 51 The analog phone is forwarded to another phone. Call the analog phone and check that the call is presented on the third phone and can be taken by this one.	OK	
2	Call to a phone, which is forwarded to another phone. The analog phone calls a phone forwarded to a third phone. The third phone takes the call and the conversation is established.	OK	
3	Forward on no answer to another phone. Prefix: 53 The analog phone is forwarded on no answer to another phone. Call the analog 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.	OK	
4	Forward on busy to another phone. Prefix: 52 The analog phone is forwarded on busy to another phone. While the analog phone is already in conversation, call the analog 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.	NOK	The call to an anlog phone already in conversation is not forwarded.

Step	Action	Result	Comment
5	Forward on busy / no answer to another phone. Prefix: 54 The analog phone is forwarded on busy / no answer to another phone. While the analog is already in conversation, call the analog 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 analog 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.	NOK	Busy: the call to an anlog phone already in conversation is not forwarded.
6	Analog phone puts call on-hold. The analog 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 analog phone. Check the conversation can be retrieved.	OK	
7	Analog phone is put on-hold. The analog phone is in conversation with another phone. The other phone puts this conversation on-hold. Check the display and on-hold music on the analog phone. Check the conversation can be retrieved.	OK	
8	Broker call. The analog phone has two active conversations and switches from one to another. Check the display and on-hold music on the phones. Check also the display and audio signalization on the analog phone.	OK	
9	Transfer in conversation. The analog 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 analog phone display (signalization of the transfer and back to idle state).	NA	Feature not avaialable on the analog phone.
10	Transfer during ringing. The analog phone has one active conversation and another one in ringing step. Before the second callee takes the call, the analog phone transfers its first call to this second callee. Check the new conversation between the two other phones is successful and also the analog phone display (signalization of the transfer and back to idle state).	OK	While second calls rings, hooking on the analog phone allows to do the transfer.

Step	Action	Result	Comment
11	Conference. The analog phone has two active conversations and initiates a conference. Check the new conversation between the three parties is successful (audio and signalization).	NA	Feature not avaialable on the analog phone.
12	Voice mail message signalization. Call the analog phone and leave a message to its voice mail (for example by forwarding the analog phone to the voice mail). Check that the message is indicated on the analog phone (led, display and/or feedback tone).	NOK	The LED on the analog phone is not ON. It only blinks one time when the message is sent. When hooking off the handset, a specific tone is played a coule of seconds to tell the user there are new messages.
13	Voice mail message listening. The analog phone has a voice mail message (see above). Press the voice mail key and interacts with the voice mail to listen to the message.	OK	
14	Voice mail message deposit. The analog phone calls another phone forwarded to the voice mail. He leaves a message. Check the interaction between the analog phone and voice mail. Listen to the message from the other phone.	OK	
15	Do Not Disturb. Prefix: 42 On the analog phone the Do not Disturb is activated. When calling this set, the call is not presented on the phone. On the analog phone the Do not Disturb is deactivated. When calling this set, the call is presented on the phone and can be picked up.	OK	
16	Wake Up. Prefix: 506 (activation) and 507 (deactivation) On the analog phone the 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 analog phone already in conversation when the wakeup time arrives. On the analog phone the Wake Up is activated then deactivated. When the previous wake up time arrives, nothing appends on the phone set.	OK	

Step	Action	Result	Comment
17	Leaving an automatic call back. Suffix: 5 The analog phone calls another phone already in conversation. The analog phone set uses the call back suffix to be recalled. Check the call back query is taken into account and processed.	OK	
18	Called by an automatic call back. Suffix: 5 The analog phone is in conversation. Another phone calls the analog phone and leaves an automatic call back. Check the call back query is taken into account and processed.	NOK	The analog phone rings, the call back is launched but when the analog phone picks up the call is not established.
19	MP118 local features. Repeat steps **1, 3, 4, 5** and **15** but this time use the MP118 local features possibilities.	OK	For the Do not disturb, MP118 sends SIP 603 Decline. The calling phone releases the call without any indication (do not disturb).

6.2.6 Fax

6.2.6.1 Test objectives

These tests check the analog fax machine (located behind the MP118) behavior during fax transmission to and from another fax machine (located in the headquarters and in the remote office).

6.2.6.2 Test procedure

Step	Action	Result	Comment
1	Fax sending to another fax machine located in the headqaurters The analog fax machine located behind the MP118 sends a fax to another fax machine located in the headquarters. Tests are performed with farious contents: simple fax, complex fax (including images), multi pages fax	OK	Tested for a single page fax.

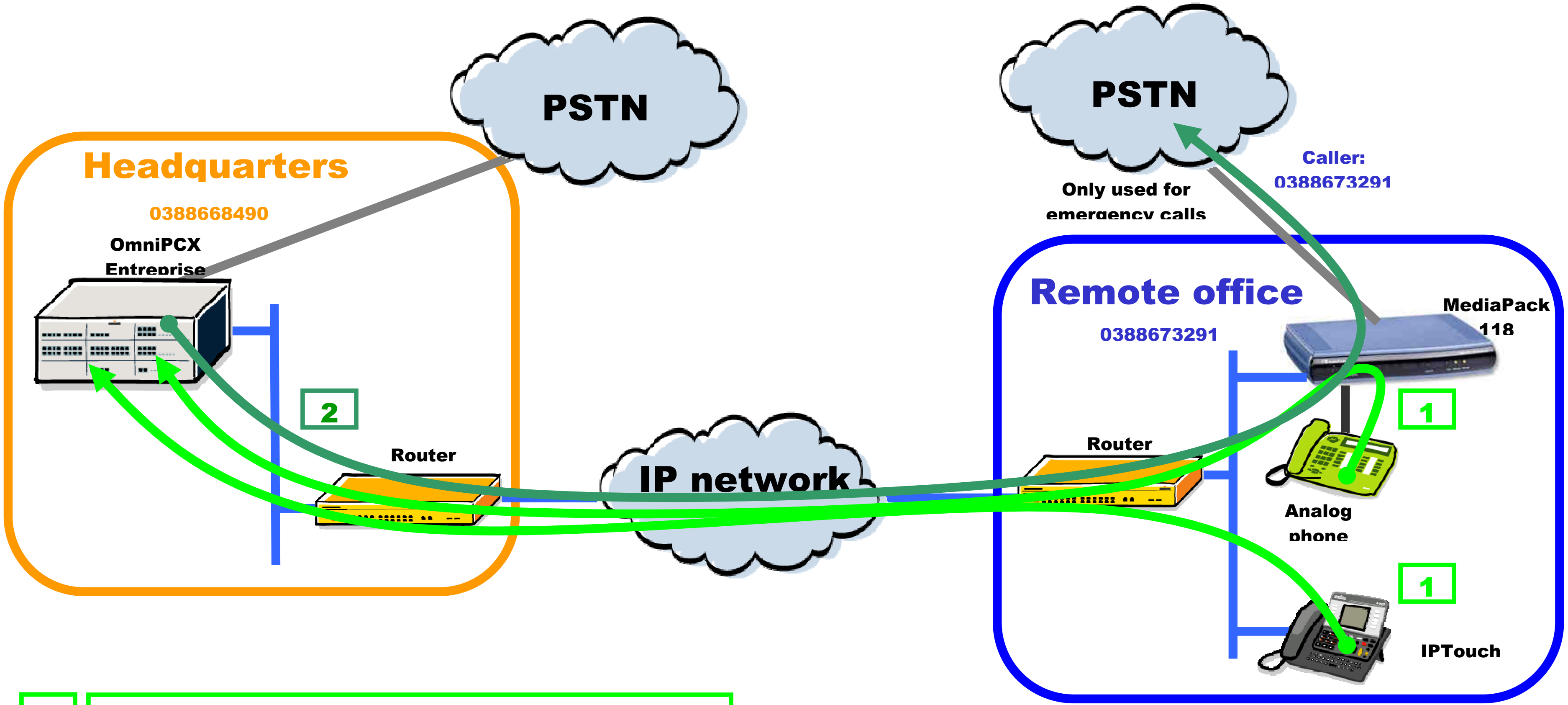
Step	Action	Result	Comment
2	Fax receiving from another fax machine located in the headqaurters The analog fax machine located behind the MP118 receives a fax from another fax machine located in the headquarters. Tests are performed with farious contents: simple fax, complex fax (including images), multi pages fax	OK	Tested for a single page fax.
3	Fax sending to a PSTN fax machine The analog fax machine located behind the MP118 sends a fax to a fax machine located in the PSTN network. Tests are performed with farious contents: simple fax, complex fax (including images), multi pages fax	OK	Tested for a single page fax.
4	Fax receiving from a PSTN fax machine The analog fax machine located behind the MP118 receives a fax from a fax machine located in the PSTN network. Tests are performed with farious contents: simple fax, complex fax (including images), multi pages fax	OK	Tested for a single page fax.

6.2.7 Emergency calls

6.2.7.1 Test objectives

These tests check the emergency calls issued by a phone located in the remote office (analog phone or IPTouch) uses the PSTN connection of the remote office to reach the public network.

In order for the emergency service to be able to localize the caller, the call has to be issued using the PSTN public network access at the closest of the caller. Thus, it is the MP118 PSTN access which is used when the caller is a phone located in the remote office.
Moreover, the caller number (sent to the emergency service) is also the remote office public network number (and not the headquarters one).



1 The analog phone or IPTouch dials the emergency number 911.
The call is routed to the OXE.

2 The OXE re-routes the call to the SIP trunk group associated with IP domain of the analog phone or IPTouch user.
The call is routed by the MP118 to PSTN public network.
Caller number is the remote office number (IP domain calling identifier).

6.2.7.2 Test procedure

Step	Action	Result	Comment
1	Analog phone connected behind the MP118 calls the emergency service Check the call is issued using the MP118 PSTN access after having been routed through the OXE first. Check the caller number is the remote office one (and not the heaquarters one).	OK	
2	IPTouch located in the remote office calls the emergency service Check the call is issued using the MP118 PSTN access after having been routed through the OXE first. Check the caller number is the remote office one (and not the heaquarters one).	OK	

6.3 MP118 used as SIP proxy (in case of IP network failure)

There is a failure in the IP network and the MP118 is used as a SIP proxy to handle the remote office IPTouch which have switched to SIP mode. Calls between the headquarters and the remote office use the PSTN network.

6.3.1 SIP registration and authentication

6.3.1.1 Test objectives

These tests check that the phones are able to register to the MP118 with and without SIP authentication.

6.3.1.2 Test procedure

Step	Action	Result	Comment
1	IPTouch in SIP mode set registration to the MP118 The IP link between the Headquarters and the remote office is cut. The phone restarts and switches to SIP mode to register to the MP118	OK	
2	IPTouch in SIP mode registration to the MP118 using SIP digest authentication Same as **1** but with SIP digest authentication.	NA	This feature does not exists in the MP118

6.3.2 Audio codec negociation

6.3.2.1 Test objectives

These tests verify that the IPTouch phone and MP118 are negociating the appropriate audio codecs for calls between the IPTouch and an analog phone located behind the MP118.

6.3.2.2 Test procedure

Step	Action	Result	Comment
1	The phone is configured to offer G711Alaw, G711 µ law, G723 and G729 (phone default behaviour which can not be changed). The MP118 is configured to use G711Alaw, G723 and G729 (in this priority order). Check that for a call from the IPTouch to the analog phone, the negociated codec is G711 Alaw Check that for a call from the analog phone to the IPTouch, the negociated codec is G711 Alaw	OK	
2	The phone is configured to offer G711Alaw, G711 µ law, G723 and G729 (phone default behaviour which can not be changed). The MP118 is configured to use G711Alaw. Check that for a call from the IPTouch to the analog phone, the negociated codec is G711 Alaw Check that for a call from the analog phone to the IPTouch, the negociated codec is G711 Alaw	OK	
3	The phone is configured to offer G711Alaw, G711 µ law, G723 and G729 (phone default behaviour which can not be changed). The MP118 is configured to use G723. Check that for a call from the IPTouch to the analog phone, the negociated codec is G723 Check that for a call from the analog phone to the IPTouch, the negociated codec is G723	OK	
4	The phone is configured to offer G711Alaw, G711 µ law, G723 and G729 (phone default behaviour which can not be changed). The MP118 is configured to use G729. Check that for a call from the IPTouch to the analog phone, the negociated codec is G729 Check that for a call from the analog phone to the IPTouch, the negociated codec is G729	OK	
5	The phone is configured to offer G711Alaw, G711 µ law, G723 and G729 (phone default behaviour which can not be changed). The MP118 is configured to use G729 and G723 (in this priority order). Check that for a call from the IPTouch to the analog phone, the negociated codec is G723 Check that for a call from the analog phone to the IPTouch, the negociated codec is G729	OK	
6	Repeat steps 1 and 2 with µ law for the MP118 configuration.	OK	

Step	Action	Result	Comment
7	No common codecs. The phone is configured to offer G711Alaw, G723 and G729. The MP118 is configured to NOT use G711Alaw, G723 and G729 (use only G726 for example). Check the behavior for a call from the IPTouch to the analog phone and for a call from the analog phone to the IPTouch.	OK	488 Not Accpetable Here are sent by the IPTouch or by the MP118

6.3.3 Defense

6.3.3.1 Test objectives

These tests check the IPTouch phone and MP118 behaviors against perturbations such as phone and MP118 reboot.

6.3.3.2 Test procedure

Step	Action	Result	Comment
1	MP118 reboot while IPTouch phone in SIP mode phone in idle. Check the phone behavior when the MP118 reboots. As soon as the MP118 is running again, the phone is able to make and receive a call.	OK	
2	MP118 reboot while IPTouch phone in SIP mode phone in conversation. Check the phone behavior while in conversation when the MP118 reboots. The call should be properly released and, as soon as the MP118 is running again, the phone is able to make and receive a call. Check for calls with an analog phone located behing the MP118, an IPTouch located in the remote office (also in SIP mode), a PSTN user.	OK	
3	IPTouch reboots while idle. Check that the phone registers again to the MP118 and is able to make and receive a call.	OK	

Step	Action	Result	Comment
4	IPTouch reboots while in conversation. Check the phone and MP118 behaviors while the IPTouch is in conversation and reboots The call should be properly released and, as soon as the IPTouch is registered again, the phone is able to make and receive a call. Check for calls with an analog phone located behing the MP118, an IPTouch located in the remote office (also in SIP mode), a PSTN user.	OK	In case of a call IPTouch to IPTouch, the phone which does not reset never releases the call. But we can imagine that the user will hang by himself. In case of a PSTN user or analog phone, the call is released.

6.3.4 Basic calls

6.3.4.1 Test objectives

These tests check the IPTouch phone and MP118 behaviors during basic calls such as IPTouch to analog phone, IPTouch to IPTouch, IPTouch to headquarters (thanks to the PSTN network), ...

6.3.4.2 Test procedure

Step	Action	Result	Comment
1	Remote office IPTouch to and from remote office IPTouch	OK	
2	Remote office IPTouch to and from remote office analog phone (behind MP118)	OK	
3	Remote office IPTouch to and from remote office SIP phone	OK	
4	Remote office IPTouch to and from headquarters phone (using PSTN network) Check with several headquarter phones: IPTouch, analog, digital, SIP.	OK	
5	Remote office analog phone and fax machine (behind MP118) to and from headquarters phone / fax machine (using PSTN network) Check with several headquarter phones: IPTouch, analog, digital, SIP.	OK	
6	Remote office IPTouch to and from PSTN user (headquarter OXE PSTN user or external PSTN user)	OK	
7	Remote office analog phone and fax machine (behind MP118) to and from PSTN external user	OK	

Step	Action	Result	Comment
8	Call release IPTouch call from and to another IPTouch both located in the remote office. IPTouch call from and to an analog phone (behind the MP118) both located in the remote office. Call is released by the called phone. Call is released by the caller.	OK	
9	Call release IPTouch located in the remote office call from and to an IPTouch located in the headquarters (using PSTN network). Analog phone (behind the MP118) located in the remote office call from and to an IPTouch located in the headquarters (using PSTN network). Call is released by the called phone. Call is released by the caller.	OK	
10	Repeat steps **8** and **9** but this time the call is released by the caller while the called phone is ringing.	OK	
11	Call rejection during ringing Incoming call to an IPTouch located in the remote office. The IPTouch rejects the call. Check when the call comes from another IPTouch located in the remote office, from an analog phone (behind MP118) located in the remote office, from an IPTouch located in the headquarters (using PSTN network), from a PSTN user.	OK	
12	Dialing break The IPTouch 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.	OK	

6.3.5 Telephonic features

6.3.5.1 Test objectives

These tests check the IPTouch phone and MP118 behaviors during telephonic feature use like forward, hold, transfer, conference, do not disturb, voice mail interactions. Programmations are done on the phone itself (IPTouch) or the MP118 (analog phones).

6.3.5.2 Test procedure

Step	Action	Result	Comment
1	IPTouch located in the remote office is forwarded (immediate forward) Check for several forward destinations: ▪ IPTouch located in the remote office, ▪ analog phone located in the remote office behind the MP118, ▪ phone located in the headquarters (using PSTN network), and for serveral callers: ▪ IPTouch located in the remote office, ▪ analog phone located in the remote office behind the MP118, ▪ phone located in the headquarters (using PSTN network), ▪ PSTN user	OK	
2	Same as **1** but this time with an analog phone located in the remote office behind the MP118.	NOK	Forward is not processed by the MP118
3	Repeat steps **1** and **2** but this time for forward on busy, forward on no answer, forward on busy / no answer.	OK but	For the IPTouch, only immediate forward is possible. For the analog phone, same as **2**
4	IPTouch located in the remote office puts remote on hold Check for several remotes: ▪ IPTouch located in the remote office, ▪ analog phone located in the remote office behind the MP118, ▪ phone located in the headquarters (using PSTN network), ▪ PSTN user.	OK but	For an analog phone, phone located in the headquartes and PSTN user there is no on hold feedback tone played. The IPTouch does not send any on hold feedback tone.
5	Same as **4** but this time with an analog phone located in the remote office behind the MP118.	OK but	For an analog phone, phone located in the headquartes and PSTN user there is no on hold feedback tone played. The IPTouch does not send any on hold feedback tone.

Step	Action	Result	Comment
6	IPTouch located in the remote office makes broker calls between two communications Check for several remotes: ▪ IPTouch located in the remote office, ▪ analog phone located in the remote office behind the MP118, ▪ phone located in the headquarters (using PSTN network), ▪ PSTN user. Check that the IPTouch can go from one conversation to another. Check also when the second call is an incoming or outgoing call.	OK	
7	Same as **6** but this time with an analog phone located in the remote office behind the MP118.	OK	
8	IPTouch located in the remote office transfers its call Check for several remotes: ▪ IPTouch located in the remote office, ▪ analog phone located in the remote office behind the MP118, ▪ phone located in the headquarters (using PSTN network), ▪ PSTN user. Check for consultative and blind transfers.	OK but	For blind transfer, the IPTouch menu (Transfer to) has to be used. When hanging on when the second call rings, it does not work. When an IPTouch calls another analog phone located behind the MP118 while already in conversation with another analog phone located behind the MP118, there is no possibility to do a blind transfer. Call has to be picked up by the analog phone to allow the IPTouch to do the transfer.
9	Same as **8** but this time with an analog phone located in the remote office behind the MP118.	NA	Feature not available on the analog phone.
10	IPTouch located in the remote office makes a 3 parties conference Check for several remotes: ▪ IPTouch located in the remote office, ▪ analog phone located in the remote office behind the MP118, ▪ phone located in the headquarters, ▪ PSTN user.	NA	Feature not available on the IPTouch.
11	Same as **10** but this time with an analog phone located in the remote office behind the MP118.	NA	Feature not available on the analog phone.

Step	Action	Result	Comment
12	IPTouch located in the remote office has activated do not disturb feature Check for several callers: ▪ IPTouch located in the remote office, ▪ analog phone located in the remote office behind the MP118, ▪ phone located in the headquarters, ▪ PSTN user.	NA	Feature not available on the IPTouch.
13	Same as **12** but this time with an analog phone located in the remote office behind the MP118.	OK but	MP118 sends SIP 603 Decline. When a phone located in the headquartersor a PSTN user calls, the call is set up (blank) and then immediately released.
14	Voice mail message deposit. The IPTouch located in the remote office 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. Check for several remotes: ▪ phone located in the headquarters (using PSTN network), ▪ PSTN user.	OK	
15	Same as **16** but this time with an analog phone located in the remote office behind the MP118.	OK	

6.3.6 Other features

6.3.6.1 Test objectives

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

6.3.6.2 Test procedure

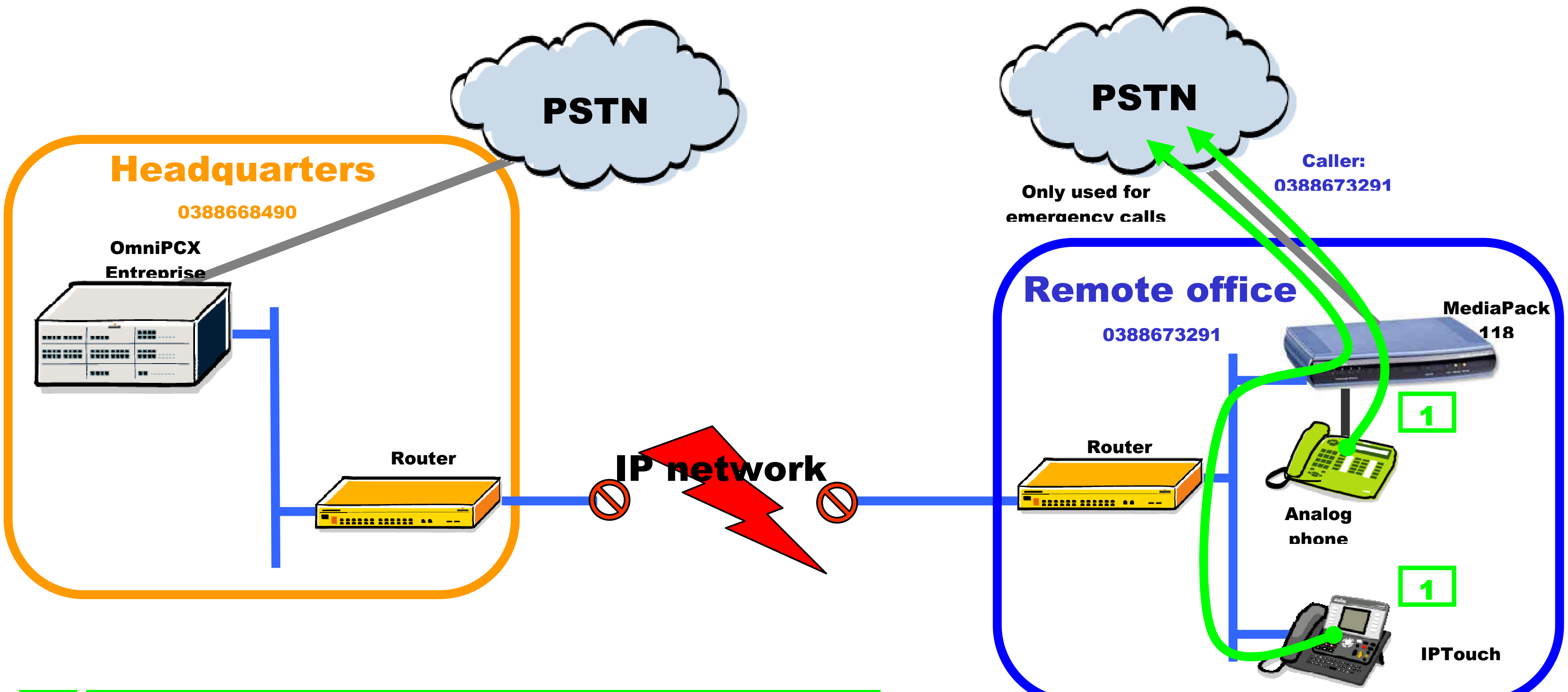
Step	Action	Result	Comment
		NT	

6.3.7 Emergency calls

6.3.7.1 Test objectives

These tests check the emergency calls issued by a phone located in the remote office (analog phone or IPTouch) use the PSTN connection of the remote office to reach the public network.

In order for the emergency service to be able to localize the caller, the call has to be issued using the PSTN public network access at the closest of the caller. Here, as the link with the headquarters is cut, , it is the MP118 PSTN access which is used when the caller is a phone located in the remote office.
Moreover, the caller number (sent to the emergency service) is also the remote office public network number.



1

The analog phone or IPTouch (in SIP mode) dials the emergency number 911.
The call is routed by the MP118 to PSTN public network.
Caller number is the remote office number.

6.3.7.2 Test procedure

Step	Action	Result	Comment
1	Analog phone connected behind the MP118 calls the emergency service Check the call is issued using the MP118 PSTN access. Check the caller number is the remote office one (and not the heaquarters one).	OK	
2	IPTouch located in the remote office calls the emergency service Check the call is issued using the MP118 PSTN access. Check the caller number is the remote office one (and not the heaquarters one).	OK	

Appendix A : Application description et configuration

Hereafter the configuration file (BOARD.ini) used for the tests.
The lines in bolded blue are explanations of the following line and can be removed from the file before loading into the Audiocodes device. These lines require special attention and must be adapted to the targeted environnement.

Note that the network parameters (IP address, network mask and gateway) of the Audiocodes device have to be configured first (see Audiocodes installation manual).

This configuration file is also available (at the same place as this interworking report) as a separate file (without the comments) directly uploadable into the Audicode device.

```
;**************
;** Ini File **
;**************

;Board: MP-118 FXS_FXO
;Serial Number: 1066289
;Slot Number: 1
;Software Version: 5.60A.025.005
;DSP Software Version: 204IM => 560.12
;Audiocodes equipement IP address
;Board IP Address: 192.168.24.105
;Audiocodes subnetwork mask
;Board Subnet Mask: 255.255.255.0
;Audiocodes default gateway IP address
;Board Default Gateway: 192.168.24.1
;Ram size: 32M   Flash size: 8M
;Num of DSP Cores: 2  Num DSP Channels: 8
;Profile: NONE
;-----------------------------------------


[SYSTEM Params]

;OXE first CPU role IP address
DNSPriServerIP = 192.168.24.6
;OXE second CPU role IP address
DNSSecServerIP = 192.168.25.6
SyslogServerIP = 192.168.24.4
VXMLFIleName = ''

[BSP Params]

PCMLawSelect = 3
;Audiocodes default gateway IP address
LocalMediaDefaultGW = 192.168.24.1
;Audiocodes equipement IP address
LocalMediaIPAddress = 192.168.24.105
;Audiocodes subnetwork mask
LocalMediaSubnetMask = 255.255.255.0
;Audiocodes equipement IP address
LocalControlIPAddress = 192.168.24.105
;Audiocodes subnetwork mask
LocalControlSubnetMask = 255.255.255.0
;Audiocodes equipement IP address
LocalOAMIPAddress = 192.168.24.105
;Audiocodes subnetwork mask
LocalOAMSubnetMask = 255.255.255.0
;Audiocodes default gateway IP address
LocalOAMDefaultGW = 192.168.24.1
StorageServerNetworkAddress = 255.255.255.255

[Analog Params]

MinFlashHookTime = 100
```

```
[ControlProtocols Params]

AdminStateLockControl = 0

[MGCP Params]


[MEGACO Params]

EP_Num_0 = 0
EP_Num_1 = 1
EP_Num_2 = 0
EP_Num_3 = 0
EP_Num_4 = 0

[Voice Engine Params]

VoiceVolume = 1
;Payload number for telephonic events (see OXE configuration: SIP / SIP gateway / Dynamic
Payload type for DTMF)
RFC2833PayloadType = 101
;Payload number for telephonic events (see OXE configuration: SIP / SIP gateway / Dynamic
Payload type for DTMF)
RFC2833TxPayloadType = 101
;Payload number for telephonic events (see OXE configuration: SIP / SIP gateway / Dynamic
Payload type for DTMF)
RFC2833RxPayloadType = 101
DTMFDetectorSensitivity = 1
TTYTRANSPORTTYPE = 1

[WEB Params]

WEBACCESSLIST_0 = 0.0.0.0
WEBACCESSLIST_1 = 0.0.0.0
WEBACCESSLIST_2 = 0.0.0.0
WEBACCESSLIST_3 = 0.0.0.0
WEBACCESSLIST_4 = 0.0.0.0
WEBACCESSLIST_5 = 0.0.0.1
WEBACCESSLIST_6 = 0.0.0.0
WEBACCESSLIST_7 = 0.0.0.0
WEBACCESSLIST_8 = 0.0.0.0
WEBACCESSLIST_9 = 0.0.0.0

[SIP Params]

ENABLECALLERID = 1
;Maximum number of digits which can be dialed on the analog phones
MAXDIGITS = 12
;Local port used by the Audiocodes equipment for the SIP signalling
LOCALSIPPORT = 7777
;Registration expiration time (in seconds) for the analog phones (see OXE configuration: SIP
/ SIP Registrar / Min expiry date and Max expiry date)
REGISTRATIONTIME = 60
;Audiocodes equipment works with a proxy (OXE)
ISPROXYUSED = 1
;Audiocodes equipment registers to the proxy (OXE)
ISREGISTERNEEDED = 1
;Audiocodes equipment waits for a dial tone before dialing (FXO) calls
ISWAITFORDIALTONE = 1
;IP calls to the Hunt Group are routed after manipulation of numbers
ROUTEMODEIP2TEL = 1
;Tel calls to IP are routed after manipulation of numbers
ROUTEMODETEL2IP = 1
;Syslog debug level (useful when investigating problems)
GWDEBUGLEVEL = 5
;Keepalive is used between tha Audiocodes equipment and the proxy (OXE): 1 means SIP OPTIONS
messages are used
ENABLEPROXYKEEPALIVE = 1
;'user=phone' is not used in SIP URI
ISUSERPHONE = 0
;OXE SIP proxy name (see OXE configuration: SIP / SIP Gateway / Machine name)
PROXYNAME = 'Etoilenoire'
;OXE SIP gateway name (see OXE configuration: SIP / SIP Gateway / Machine name)
SIPGATEWAYNAME = 'Etoilenoire'
CNONCE = '0a123bcf'
PASSWORD = '787899'
;Tel to IP routing table is used in case proxys servers (OXE) unavailability
ISFALLBACKUSED = 1
;Audiocodes equipment will send INVITE/REGISTER to next redundant proxy in case the first
one does not answer
ISPROXYHOTSWAP = 1
```

```
;Time (in seconds) between two keeapalive messages to the proxy (OXE)
PROXYKEEPALIVETIME = 10
;Alternate routing is not used
ALTROUTINGTEL2IPMODE = 0
;Calls are not released in case of RTP packets are no more received
DISCONNECTONBROKENCONNECTION = 0
;Subscription to the voice mail message waiting indication
ENABLEMWISUBSCRIPTION = 1
;Voice mail (see OXE 4645 configuration)
MWISERVERIP = 'Etoilenoire'
MWIANALOGLAMP = 1
;Voice mail message waiting indication service is activated
ENABLEMWI = 1
;Audiocodes equipment checks the routing rules in the 'Tel to IP
Routing' table for a match with the Tel-to-IP call. Only if a match is
not found is a Proxy used
PREFERROUTETABLE = 1
;Minimum registration expiration time (in seconds) accepted for the IPTouch which connect to
the Audiocodes equipment
MINSE = 180
;T38 is used for SIP signalling methof for fax sessions
ISFAXUSED = 1
;UPDATE method is used for session-timer updates
SESSIONEXPIRESMETHOD = 1
;Audiocodes equipement SAS module IP address and port number
SASDEFAULTGATEWAYIP = '192.168.24.105:7777'
;Stand alone survivability is activated
ENABLESAS = 1
;Stand alone survivability port used
SASLOCALSIPUDPPORT = 5060
;UDP is used for SIP dialogs with the voice mail
MWISERVERTRANSPORTTYPE = 0
;Proxy Set (index number) used in SAS Normal mode to forward REGISTER and INVITE requests.
See also [ProxyIP] section
SASPROXYSET = 1
;Only the user part is used for the SAS database binding
SASBINDINGMODE = 1
;The FXO device plays a tone to the TDM side if a Tel-to-IP call is rejected by a SIP error
response (4xx, 5xx or 6xx).
FXOAutoDialPlayBusyTone = 1

[IPsec Params]


[SNMP Params]

DisableSNMP = 1

;
;  *** TABLE DspTemplates ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts
;


;
;  *** TABLE InterfaceTable ***
;
;


;Audiocodes equipement IP parameters
[ InterfaceTable ]
FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes, InterfaceTable_InterfaceMode,
InterfaceTable_IPAddress, InterfaceTable_PrefixLength, InterfaceTable_Gateway,
InterfaceTable_VlanID, InterfaceTable_InterfaceName;
InterfaceTable 0 = 6, 10, 192.168.24.105, 24, 192.168.24.1, 1, O+M+C;

[ \InterfaceTable ]

;
;  *** TABLE CoderName ***
;
;


;Audiocodes VoIP coders list
[ CoderName ]
FORMAT CoderName_Index = CoderName_Type, CoderName_PacketInterval, CoderName_rate,
CoderName_PayloadType, CoderName_Sce;
CoderName 0 = g711Alaw64k, 20, 0, 255, 0;
CoderName 1 = g7231, 30, 0, 255, 0;
CoderName 2 = g729, 20, 0, 255, 0;
CoderName 5 = g711Alaw64k, 20, 0, 255, 0;
```

```
CoderName 6 = g7231, 30, 0, 255, 0;
CoderName 7 = g729, 20, 0, 255, 0;
CoderName 10 = g711Alaw64k, 20, 0, 255, 0;
CoderName 15 = g7231, 30, 0, 255, 0;
CoderName 20 = g729, 20, 0, 255, 0;

[ \CoderName ]

;
;  *** TABLE TrunkGroup ***
;
;

;Audiocodes equipments endpoints and hunt group assignements
;22018 is analog phone connected on FXS port 1
;22019 is analog phone connected on FXS port 2
;0 is the PSTN access prefix (FXO port 5)
;26005 is analog fax connected on FXS port 4
;22020 is analog phone connected on FXS port 3

[ TrunkGroup ]
FORMAT TrunkGroup_Index = TrunkGroup_TrunkGroupNum, TrunkGroup_FirstTrunkId,
TrunkGroup_FirstBChannel, TrunkGroup_LastBChannel, TrunkGroup_FirstPhoneNumber,
TrunkGroup_ProfileId, TrunkGroup_LastTrunkId, TrunkGroup_Module;
TrunkGroup 0 = 1, 255, 1, 1, 22018, 1, 255, 255;
TrunkGroup 1 = 1, 255, 2, 2, 22019, 1, 255, 255;
TrunkGroup 2 = 3, 255, 5, 5, 0, 1, 255, 255;
TrunkGroup 3 = 1, 255, 4, 4, 26005, 1, 255, 255;
TrunkGroup 4 = 1, 255, 3, 3, 22020, 1, 255, 255;

[ \TrunkGroup ]

;
;  *** TABLE NumberMapIp2Tel ***
;
;

;Manipulation of destination number of IP to Tel calls: PSTN access prefix 0 and emergency
call prefix 911
[ NumberMapIp2Tel ]
FORMAT NumberMapIp2Tel_Index = NumberMapIp2Tel_DestinationPrefix,
NumberMapIp2Tel_SourcePrefix, NumberMapIp2Tel_SourceAddress, NumberMapIp2Tel_NumberType,
NumberMapIp2Tel_NumberPlan, NumberMapIp2Tel_RemoveFromLeft, NumberMapIp2Tel_RemoveFromRight,
NumberMapIp2Tel_LeaveFromRight, NumberMapIp2Tel_Prefix2Add, NumberMapIp2Tel_Suffix2Add,
NumberMapIp2Tel_IsPresentationRestricted, NumberMapIp2Tel_SrcTrunkGroupID,
NumberMapIp2Tel_SrcIPGroupID;
NumberMapIp2Tel 1 = 0, *, *, 255, 255, 1, 0, 255, , , 255, -1, -1;
NumberMapIp2Tel 2 = 911, *, *, 255, 255, 0, 0, 255, , , 255, -1, -1;

[ \NumberMapIp2Tel ]

;
;  *** TABLE PstnPrefix ***
;
;

;Routing of IP to Tel calls to hunt groups: emergency call prefix 911 and PSTN access prefix
0
[ PstnPrefix ]
FORMAT PstnPrefix_Index = PstnPrefix_DestPrefix, PstnPrefix_TrunkGroupId,
PstnPrefix_SourcePrefix, PstnPrefix_SourceAddress, PstnPrefix_ProfileId,
PstnPrefix_SrcIPGroupID, PstnPrefix_DestHostPrefix, PstnPrefix_SrcHostPrefix;
PstnPrefix 0 = 911, 3, *, *, 1, -1, , ;
PstnPrefix 1 = 0, 3, *, *, 1, -1, , ;
PstnPrefix 3 = *, 1, *, *, 1, -1, , ;
PstnPrefix 4 = *, 2, *, *, 1, -1, , ;
PstnPrefix 5 = *, 4, *, *, 1, -1, , ;

[ \PstnPrefix ]

;
;  *** TABLE Dns2Ip ***
;
;

;Audiocodes equipment DNS table (containing both OXE Call Servers main role IP address)
[ Dns2Ip ]
FORMAT Dns2Ip_Index = Dns2Ip_DomainName, Dns2Ip_FirstIpAddress, Dns2Ip_SecondIpAddress,
Dns2Ip_ThirdIpAddress, Dns2Ip_FourthIpAddress;
Dns2Ip 0 = Etoilenoire, 192.168.24.6, 192.168.25.6, 0.0.0.0, 0.0.0.0;

[ \Dns2Ip ]
```

```
;Proxys list
;Proxy 0 is the Audicodes equipment itself
;Proxy 1 is the OXE (with the two Call Servers role addresses as proxy servers)
;
;  *** TABLE ProxyIp ***
;
;

[ ProxyIp ]
FORMAT ProxyIp_Index = ProxyIp_IpAddress, ProxyIp_TransportType, ProxyIp_ProxySetId;
ProxyIp 0 = 192.168.24.105:5060, -1, 0;
ProxyIp 1 = 192.168.24.6, -1, 1;
ProxyIp 2 = 192.168.25.6, -1, 1;

[ \ProxyIp ]

;
;  *** TABLE TxDtmfOption ***
;
;

;RFC2833 is used to carry DTMF events
[ TxDtmfOption ]
FORMAT TxDtmfOption_Index = TxDtmfOption_Type;
TxDtmfOption 0 = 4;

[ \TxDtmfOption ]

;
;  *** TABLE TrunkGroupSettings ***
;
;

;Hunt groups channel selection mode
[ TrunkGroupSettings ]
FORMAT TrunkGroupSettings_Index = TrunkGroupSettings_TrunkGroupId,
TrunkGroupSettings_ChannelSelectMode, TrunkGroupSettings_RegistrationMode,
TrunkGroupSettings_GatewayName, TrunkGroupSettings_ContactUser,
TrunkGroupSettings_ServingIPGroup;
TrunkGroupSettings 0 = 1, 0, 0, , , -1;
TrunkGroupSettings 1 = 2, 2, 0, , , -1;
TrunkGroupSettings 2 = 3, 1, 0, , , -1;
TrunkGroupSettings 3 = 4, 0, 0, , , -1;

[ \TrunkGroupSettings ]

;
;  *** TABLE TelProfile ***
;
;

;Telephone profiles
[ TelProfile ]
FORMAT TelProfile_Index = TelProfile_ProfileName, TelProfile_TelPreference,
TelProfile_CodersGroupID, TelProfile_IsFaxUsed, TelProfile_JitterBufMinDelay,
TelProfile_JitterBufOptFactor, TelProfile_IPDiffServ, TelProfile_SigIPDiffServ,
TelProfile_DtmfVolume, TelProfile_InputGain, TelProfile_VoiceVolume,
TelProfile_EnableReversePolarity, TelProfile_EnableCurrentDisconnect,
TelProfile_EnableDigitDelivery, TelProfile_EnableEC, TelProfile_MWIAnalog,
TelProfile_MWIDisplay, TelProfile_FlashHookPeriod, TelProfile_EnableEarlyMedia,
TelProfile_ProgressIndicator2IP, TelProfile_TimeForReorderTone, TelProfile_EnableDIDWink,
TelProfile_IsTwoStageDial, TelProfile_DisconnectOnBusyTone;
TelProfile 1 = , 1, 0, 1, 10, 10, 46, 40, -11, 0, 1, 0, 0, 0, 1, 1, 0, 700, 1, 0, 255, 0, 0,
1;
TelProfile 2 = , 1, 2, 1, 10, 10, 46, 40, -11, 0, 1, 0, 0, 0, 1, 1, 1, 700, 0, -1, 255, 0,
1, 1;

[ \TelProfile ]

;
;  *** TABLE IpProfile ***
;
;

;IP profiles
[ IpProfile ]
FORMAT IpProfile_Index = IpProfile_ProfileName, IpProfile_IpPreference,
IpProfile_CodersGroupID, IpProfile_IsFaxUsed, IpProfile_JitterBufMinDelay,
IpProfile_JitterBufOptFactor, IpProfile_IPDiffServ, IpProfile_SigIPDiffServ, IpProfile_SCE,
IpProfile_RTPRedundancyDepth, IpProfile_RemoteBaseUDPPort, IpProfile_CNGmode,
IpProfile_VxxTransportType, IpProfile_NSEMode, IpProfile_IsDTMFUsed,
```

```
IpProfile_PlayRBTone2IP, IpProfile_EnableEarlyMedia, IpProfile_ProgressIndicator2IP,
IpProfile_EnableEchoCanceller, IpProfile_CopyDest2RedirectNumber,
IpProfile_MediaSecurityBehaviour, IpProfile_CallLimit,
IpProfile_DisconnectOnBrokenConnection, IpProfile_EnableHold, IpProfile_InputGain,
IpProfile_VoiceVolume;
IpProfile 0 = , 1, 0, 1, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0, 0, -1, 1, 0, 0, -1, 0, 1, -
1, 1;
IpProfile 1 = , 1, 0, 1, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 1, 1, 0, 1, 0, 0, -1, 0, 1, -
1, 1;

[ \IpProfile ]

;Caller display (caller ID)
;Analog phone connected to FXS port 1 name is 'Benoit Martin'
;Analog phone connected to FXS port 2 name is 'David Cayer'
;Analog phone connected to FXS port 3 name is 'Miroslav Stolc'
;Analog fax connected to FXS port 4 name is 'Fax'
;PSTN access (FXO port 5) name is 'Public'
;
;  *** TABLE CallerDisplayInfo ***
;
;

[ CallerDisplayInfo ]
FORMAT CallerDisplayInfo_Index = CallerDisplayInfo_DisplayString,
CallerDisplayInfo_IsCidRestricted;
CallerDisplayInfo 0 = "Benoit Martin", 0;
CallerDisplayInfo 1 = "David Cayer", 0;
CallerDisplayInfo 2 = "Miroslav Stolc", 0;
CallerDisplayInfo 3 = Fax, 0;
CallerDisplayInfo 4 = Public, 0;

[ \CallerDisplayInfo ]



;
;  *** TABLE TargetOfChannel ***
;
;

;Number automatic dialed. Analog phone 22018 is rung when there is an incoming call on the
PSTN acces (FXO port 5)
[ TargetOfChannel ]
FORMAT TargetOfChannel_Index = TargetOfChannel_Destination, TargetOfChannel_Type;
TargetOfChannel 4 = 22018, 1;

[ \TargetOfChannel ]

;
;  *** TABLE ProxySet ***
;
;

;Proxy sets paramaters
;Proxy 0 (Audiocodes equipment) uses no keepalive and is not hot swap
;Proxy 1 (OXE) uses keepalive and is hot swap
[ ProxySet ]
FORMAT ProxySet_Index = ProxySet_EnableProxyKeepAlive, ProxySet_ProxyKeepAliveTime,
ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap;
ProxySet 0 = 0, 10, 0, 0;
ProxySet 1 = 1, 10, 0, 1;

[ \ProxySet ]

;Logical IP entities used in the call routing tables
;
;  *** TABLE IPGroup ***
;
;

[ IPGroup ]
FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Description, IPGroup_ProxySetId,
IPGroup_SIPGroupName, IPGroup_ContactUser, IPGroup_EnableSurvivability,
IPGroup_ServingIPGroup, IPGroup_SipReRoutingMode, IPGroup_AlwaysUseRouteTable,
IPGroup_RoutingMode;
IPGroup 1 = 0, OXE, 1, , , 0, -1, 0, 0, -1;
IPGroup 2 = 0, MP118, 0, , , 0, -1, 0, 0, -1;
IPGroup 3 = 0, , 1, , , 0, -1, 0, 0, -1;
IPGroup 4 = 0, , 1, , , 0, -1, 0, 0, -1;
IPGroup 5 = 0, , 1, , , 0, -1, 0, 0, -1;
```

```
IPGroup 6 = 0, , 1, , , 0, -1, 0, 0, -1;
IPGroup 7 = 0, , 1, , , 0, -1, 0, 0, -1;
IPGroup 8 = 0, , 1, , , 0, -1, 0, 0, -1;
IPGroup 9 = 0, , 1, , , 0, -1, 0, 0, -1;

[ \IPGroup ]

;
;  *** TABLE SASRegistrationManipulation ***
;
;

;Used by the SAS application to manipulate the User-Part of an incoming REGISTER request
[ SASRegistrationManipulation ]
FORMAT SASRegistrationManipulation_Index = SASRegistrationManipulation_RemoveFromRight,
SASRegistrationManipulation_LeaveFromRight;
SASRegistrationManipulation 0 = 0, 0;

[ \SASRegistrationManipulation ]
```

Appendix B: Alcatel-Lucent Communication Platform: configuration requirements

1. Remote office analog phone configuration:

```
Consult/Modify: Users

            Node Number (reserved) : 102
                  Directory Number : 22018

                    Directory name : Martin
              Directory First Name : Benoit
                          Set Type + SIP extension
                Voice Mail Dir.No. : 20000
```

```
Consult/Modify: IP SIP Extension

            Node Number (reserved) : 102
                  Directory Number : 22018
                  Directory Number : 22018

                          Set Type + SIP extension
                        IP Address : 192.168.24.105
```

2. Remote office analog fax configuration:

```
Consult/Modify: Users

            Node Number (reserved) : 102
                  Directory Number : 26005

                    Directory name : Fax3
              Directory First Name : --------------------
                          Set Type + SIP device
```

```
Consult/Modify: IP SIP Extension

            Node Number (reserved) : 102
                  Directory Number : 26005
                  Directory Number : 26005

                          Set Type + SIP device
                        IP Address : 192.168.24.105
```

3. Remote office IP domain configuration:

```
Consult/Modify: IP Domain

            Node Number (reserved) : 102
               Instance (reserved) : 1
                  IP Domain Number : 1

                    IP Domain Name : ROAudiocodes
                    Trunk Group Id : 5
                Calling Identifier : 0388673291
    Supplement. Calling Identifier : ------------------------------
            SIP Survivability Mode + YES
                    IP Domain Type + IP
                   SIP DNS Address : 192.168.24.105
                   SIP Domain Name : 192.168.24.105
          SIP Authentication Realm : Realm1
                  Proxy IP Address : 192.168.24.105
                 Proxy Port Number : 5060
                SIP Transport Mode + UDP
```

```
┌Consult/Modify: IP Domain Address──────────────────────┐
│                                                        │
│          Node Number (reserved) : 102                  │
│             Instance (reserved) : 1                    │
│                IP Domain Number : 1                    │
│                  IP Address Low : 192.168.24.105       │
│                                                        │
│                 IP Address High : 192.168.24.109       │
│                      NetMask IP : 255.255.255.0        │
│                 IP Address Type + IP Range             │
│                                                        │
└────────────────────────────────────────────────────────┘
```

4. <u>Remote office IPTouch configuration:</u>

```
┌Consult/Modify: Users─────────────────────────────┐
│            Node Number (reserved) : 102          │
│                  Directory Number : 21013        │
│                    Directory name : Dirnbach     │
│              Directory First Name : Milan        │
│                          Set Type + IPTouch 4008 │
│                Voice Mail Dir.No. : 20000        │
│                SIP Authentication : 21013        │
│                          Password : ****         │
│                           Confirm : ****         │
└──────────────────────────────────────────────────┘
┌Consult/Modify: Tsc Ip User─────────────────────────────┐
│                                                        │
│            Node Number (reserved) : 102                │
│                  Directory Number : 21013              │
│                  Directory Number : 21013              │
│                                                        │
│                          Set Type + IPTouch 4008       │
│                        IP Address : 192.168.24.106     │
│                  IP Domain Number : 1                  │
│                                                        │
└────────────────────────────────────────────────────────┘
```

5. <u>Internal SIP gateway configuration:</u>

```
┌Consult/Modify: SIP Gateway───────────────────────────┐
│                                                      │
│            Node Number (reserved) : 102              │
│               Instance (reserved) : 1                │
│               Instance (reserved) : 1                │
│                                                      │
│                 Subnetwork number : 2                │
│                       Trunk Group : 2                │
│                        IP Address : 192.168.24.6     │
│                       Machin name : Etoilenoire      │
│                 Proxy Port Number : 5060             │
│        SIP Subscribe Min Duration : 60               │
│             DNS local domain name : etoilenoire.fr   │
│     Dynamic Payload type for dtmf : 101              │
│                                                      │
└──────────────────────────────────────────────────────┘
```

6. <u>Internal SIP proxy configuration:</u>

```
┌Consult/Modify: Proxy──────────────────────────────┐
│                                                   │
│            Node Number (reserved) : 102           │
│               Instance (reserved) : 1             │
│               Instance (reserved) : 1             │
│                                                   │
│     Minimal authentication method + None          │
│              Authentication realm : icehockey     │
│ Only authenticated incoming calls + False         │
│                                                   │
└───────────────────────────────────────────────────┘
```

7. Internal SIP registrar configuration:

```
Consult/Modify: Registrar

             Node Number (reserved) : 102
                Instance (reserved) : 1
                Instance (reserved) : 1

                    Min expiry date : 60
```

8. IP domain configuration check:

```
(102)Iceberg> domstat

Mon Aug 31 14:39:56 CEST 2009


DISPLAY DOMAIN INFO Menu
-------------------------------------
Display one Domain                      :1
Display all Domains                     :2
Display all survivability SIP Domains   :3
Display one Entry                       :4
Display all Domains Entries             :5
Display one Domain Entries              :6
Display IP Hash Table                   :7
Display one Domain Devices              :8
Display all Domains Devices             :9
Display one Domain In Service Devices   :10
Display all Domains In Service Devices  :11
Quit this tool                          :0

Enter your choice : 3
+-------+-----------+--------+---------+---------+---------+----------+--------+-----------
+--------------------++-----------------+
|Dom Id.|  Dom Type  |Survival|Max. Cnx.|Ext. Algo|Int. Algo|QoS Categ.|Trk.Grp.|Qos IP
Rec.|    Time Zone Name  |   Country Name  |
+-------+-----------+--------+---------+---------+---------+----------+--------+-----------
+--------------------++-----------------+
|      1|   IP_REMOTE|  SIP   |       -1|     G711|     G711|         0|       5|
0|        Europe/Paris|              FRA|
+-------+-----------+--------+---------+---------+---------+----------+--------+-----------
+--------------------++-----------------+
| Calling Number     : *38867329100000000000000000000|
| Sup Calling Number : 000000000000000000000000000000 |
+-------+-----------+--------+---------+---------+---------+----------+--------+-----------
+--------------------++-----------------+
| SIP domain name  : 192.168.24.105
| DNS address      : 192.168.024.105
| Authentification : Realm1
| Proxy address    : 192.168.24.105
| Proxy port       : 5060
| Transport mode   : UDP
| SIP diffserv     : 40
+-------+-----------+--------+---------+---------+---------+----------+--------+-----------
+--------------------++-----------------+

return to menu : press  ENTER


Enter your choice : 6
Enter Domain Id:1
+----+----+----+----------------+----------------+-----------------+----+----+
| Id.|Dom.|Type| Low IP Address  | High IP Address |  IP Net Mask    |H  D|H SN|
+----+----+----+----------------+----------------+-----------------+----+----+
|   0|   1|  R.|  192.168.024.105|  192.168.024.109|  255.255.255.000|  -1|   1|
+----+----+----+----------------+----------------+-----------------+----+----+



Enter your choice : 8
Enter Domain Id:1
```

```
----------------------IP couplers defined in domain   1 IP_REMOTE-------------
    NOTHING

----------------------IP Terminals in domain 1 IP_REMOTE --------------------
+--------+-------------+------------------+-+-----+---------------+----+---+
| QMCDU  | Name        |    Mac Address   | |Neqt |  Ip Address   |Type|   |
+--------+-------------+------------------+-+-----+---------------+----+---+
|21013   |Dirnbach Milan|00:80:9f:6b:6c:87   |V|01260|192.168.024.106| Ipt|  E|
|21028   |Resetka Pavol |00:80:9f:6b:6e:46   |V|01261|192.168.024.107| Ipt|  E|
+--------+-------------+------------------+-+-----+---------------+----+---+

        In the first single column
        V: means the channel is valid
        -: means the channel is not valid

        In the second column
        L: means the set is locked for binary download
        G: means the set is gigabit
        E: means the set is extended edition


----------------------sip extension terminals in domain 1 IP_REMOTE ---------
+--------+-----+---------------+--------------+-----+
| QMCDU  |Neqt |     Name      |  Ip Address   |State|
+--------+-----+---------------+--------------+-----+
|22018   |01255|Martin Benoit  |192.168.024.105| ES  |
|22019   |01259|Cayer David    |192.168.024.105| ES  |
|22020   |01280|Stolc Miroslav |192.168.024.105| ES  |
+--------+-----+---------------+--------------+-----+

----------------------sip devices in domain 1 IP_REMOTE ---------------------
+--------+----------------+---------------+
| QMCDU  |     Name       |  Ip Address   |
+--------+----------------+---------------+
|26005   |Fax3            |192.168.024.105|
+--------+----------------+---------------+
```

9. Emergency call & SIP trunking configurations:

Headquesters entity

```
┌Consult/Modify: Entities──────────────────────────────┐
│                                                      │
│             Node Number (reserved) : 102             │
│                      Entity Number : 1               │
│                                                      │
│                               Name : ENTITY_1        │
│             Installation No (ISDN) : 0388668490      │
│       Supplement.Install.No (ISDN) : ------------    │
│                     Trunk Group Id : 0               │
└──────────────────────────────────────────────────────┘
```

Remote office entity: remote office installation number used when issuing an emergency call.
The trunk group associated with this domaon is the SIP trunk to the Audiocodes

```
┌Consult/Modify: Entities────────────────────────────────┐
│                                                        │
│             Node Number (reserved) : 102               │
│                      Entity Number : 2                 │
│                                                        │
│                               Name : AudiocodesRO      │
│             Installation No (ISDN) : 0388668490        │
│       Supplement.Install.No (ISDN) : ------------      │
│                     Trunk Group Id : 5                 │
└────────────────────────────────────────────────────────┘
```

Categories / Access Category

```
┌Consult/Modify: Access Category───────────────────────┐
│                                                      │
│             Node Number (reserved) : 102             │
│                Instance (reserved) : 1               │
│            Public Network Category : 2               │
│                                                      │
│                                ARS privilege         │
│                                                      │
```

```
                    Night : 20
                      Day : 20
                   Mode 1 : 20
                   Mode 2 : 20
```

Translator / Prefix plan: emergency call is 911. 91 is the ARS prefix

```
┌Consult/Modify: Prefix Plan─────────────────────────────────────┐
│                                                                  │
│          Node Number (reserved) : 102                            │
│             Instance (reserved) : 1                              │
│                          Number : 91                             │
│                                                                  │
│                  Prefix Meaning : ARS Prof.Trg Grp Seizure       │
│               Discriminator Nr. : 6                              │
│                                                                  │
└──────────────────────────────────────────────────────────────────┘
```

Translator / External numbering plan / Numbering discriminator / Discriminator rule:
emergency call is 911. 91 is the ARS prefix and 1 is the additional called number

```
┌Consult/Modify: Discriminator Rule─────────┐
│                                            │
│          Node Number (reserved) : 102      │
│             Instance (reserved) : 1        │
│             Instance (reserved) : 1        │
│               Discriminator Nr. : 4        │
│                     Call Number : 1        │
│                                            │
│           ARS Route List Number : 24       │
│                Number of Digits : 1        │
│                                            │
└────────────────────────────────────────────┘
```

Translator / Automatic route selection / ARS route list

```
┌Consult/Modify: ARS Route list─────────────────┐
│                                                │
│          Node Number (reserved) : 102          │
│             Instance (reserved) : 1            │
│             Instance (reserved) : 1            │
│                  ARS Route list : 24           │
│                                                │
│                            Name : emergency    │
│                                                │
└────────────────────────────────────────────────┘
```

Translator / Automatic route selection / ARS route list / ARS route: 911 wil be sent to the
public network. The trunk group used is the trunk group of the IP domain of the caller.

```
┌Consult/Modify: ARS Route───────────────────────────────────────────┐
│                                                                      │
│          Node Number (reserved) : 102                                │
│             Instance (reserved) : 1                                  │
│             Instance (reserved) : 1                                  │
│                  ARS Route list : 24                                 │
│                           Route : 1                                  │
│                                                                      │
│                            Name : emergency                          │
│              Trunk Group Source : IP Domain                          │
│                     Trunk Group : -1                                 │
│         Nb.Digits To Be Removed : 1                                  │
│                   Digits To Add : 911                                │
│       Numbering Command Tabl.Id : 38                                 │
│                  NPD identifier : 34                                 │
│                      Route Type : Public                             │
│                                                                      │
│                                Quality                               │
│                                                                      │
│         [  Add   ]    [ Remove ]    [  Next  ]    [Previous]         │
│                                                                      │
│                      Quality : Speech                                │
│                                                                      │
```

Translator / External numbering plan / Numbering plan description (NPD): the installation number used is the IP domain one

```
┌Consult/Modify: Numbering Plan Description (NPD)──────┐
│                                                       │
│           Node Number (reserved) : 102                │
│              Instance (reserved) : 1                  │
│              Instance (reserved) : 1                  │
│           Description identifier : 34                 │
│                                                       │
│                             Name : emergency          │
│           Install. number source : IP Domain source   │
│            Default number source : IP Domain source   │
│                                                       │
└───────────────────────────────────────────────────────┘
```

Translator / Automatic route selection / ARS route list / Time based route list

```
┌Consult/Modify: Time Based Route List────────────────────────────┐
│                                                                  │
│           Node Number (reserved) : 102                           │
│              Instance (reserved) : 1                             │
│              Instance (reserved) : 1                             │
│                   ARS Route list : 24                            │
│         Time Based Route List Id : 1                             │
│                                                                  │
│                             Time Based Route                     │
│                                                                  │
│          [  Add   ]    [ Remove ]    [  Next  ]    [Previous]    │
│                                                                  │
│                             Time Based Route                     │
│                                                                  │
│                     Route Number : 1                             │
│               Waiting Cost Limit : 0                             │
│              Stopping Cost Limit : 0                             │
│                                                                  │
└──────────────────────────────────────────────────────────────────┘
```

Translator / Automatic route selection / Numbering command table

```
┌Consult/Modify: Numbering Command Table────────┐
│                                                │
│           Node Number (reserved) : 102         │
│              Instance (reserved) : 1           │
│              Instance (reserved) : 1           │
│                         Table Id : 37          │
│                                                │
│                Carrier Reference : 0           │
│                          Command : I           │
│           Associated SIP gateway : –1          │
│                                                │
└────────────────────────────────────────────────┘
```

Trunk groups: the SIP trunk group to the Audiocodes FXO port. Used for the emergency call.

```
┌Consult/Modify: Trunk Groups─────────────────────────────────────┐
│                                                                  │
│                 Trunk Group Name : SIP Audioc                    │
│           Number Compatible With : –1                            │
│                   Remote Network : 12                            │
│                      Node number : 2                             │
│              Q931 signal variant : ISDN all countries            │
│         Number Of Digits To Send : 0                             │
│                   T2 Specificity : SIP                           │
│          Public Network Category : 31                            │
│                                                                  │
└──────────────────────────────────────────────────────────────────┘
```

Trunk groups / Trunk group

```
┌Consult/Modify: Trunk Group───────────────────────┐
│                                                   │
│           Node Number (reserved) : 102            │
│                   Trunk Group Id : 5              │
```

```
              Instance (reserved) : 1

                  Trunk Group Type : T2
                   T2 Specificity : SIP
                    Entity Number : 2
```

Translator / Network routing table: the SIP external gateway associated to the SIP trunk

```
┌Consult/Modify: Network Routing Table

           Node Number (reserved) : 102
              Instance (reserved) : 1
                   Network Number : 12

     Rank of First Digit to be Sent : 1
                    Protocol Type : ABC_F
      Numbering Plan Descriptor Id : 11
                   ARS Route list : 0
           Associated SIP gateway : 4
```

SIP / External gateways:

```
┌Consult/Modify: External Gateways

           Node Number (reserved) : 102
              Instance (reserved) : 1
                         Instance : 4

                     Gateway Name : Audiocodes
                    Remote domain : 192.168.24.105
                      Port number : 5060
                   Transport type : TCP
               Belonging  domain : etoilenoire.fr
                  Registration Id : ____________________
                Trunk group number : 5
```

Translator / Automatic route selection / ARS route list / ARS route

```
┌Consult/Modify: ARS Route

           Node Number (reserved) : 102
              Instance (reserved) : 1
              Instance (reserved) : 1
                   ARS Route list : 25
                            Route : 1

                             Name : Audiocdes
               Trunk Group Source : Route
                      Trunk Group : 5
          Nb.Digits To Be Removed : 0
                    Digits To Add : ______________________________
        Numbering Command Tabl.Id : 38
                   NPD identifier : 35
                       Route Type : Public

                               Quality

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

                          Quality ┌ Speech
```

Translator / External numbering plan / Numbering plan description (NPD) : the installation number used is the IP domain one

```
┌Consult/Modify: Numbering Plan Description (NPD)────────┐
│                                                        │
│            Node Number (reserved) : 102                │
│               Instance (reserved) : 1                  │
│               Instance (reserved) : 1                  │
│            Description identifier : 35                 │
│                                                        │
│                              Name : Audiocodes         │
│            Install. number source : IP Domain source   │
│             Default number source : IP Domain source   │
│                                                        │
└────────────────────────────────────────────────────────┘
```

```
Translator / Automatic route selection / ARS route list / Time based route list
```

```
┌Consult/Modify: Time Based Route List──────────────────────────┐
│                                                                │
│            Node Number (reserved) : 102                        │
│               Instance (reserved) : 1                          │
│               Instance (reserved) : 1                          │
│                    ARS Route list : 25                         │
│          Time Based Route List Id : 1                          │
│                                                                │
│                             Time Based Route                   │
│                                                                │
│           [  Add   ]    [ Remove ]    [  Next  ]    [Previous] │
│                                                                │
│                             Time Based Route                   │
│                                                                │
│                      Route Number : 1                          │
│                Waiting Cost Limit : 0                          │
│               Stopping Cost Limit : 0                          │
│                                                                │
└────────────────────────────────────────────────────────────────┘
```

```
Translator / Automatic route selection / Numbering command table
```

```
┌Consult/Modify: Numbering Command Table──────────────────────┐
│                                                             │
│            Node Number (reserved) : 102                     │
│               Instance (reserved) : 1                       │
│               Instance (reserved) : 1                       │
│                          Table Id : 38                      │
│                                                             │
│                 Carrier Reference : 0                       │
│                           Command : I                       │
│            Associated SIP gateway : 4                       │
│                                                             │
└─────────────────────────────────────────────────────────────┘
```

Appendix C: Partner escalation process

In case you would need technical assistance, please contact the reseller/distributor where you purchased your AudioCodes products. They have been trained on the products to give you 1st and 2nd levels of support. They are in plus in direct relation with 3rd level AudioCodes support in case an escalation would be needed.

Appendix D: AAPP program, documentation and technical assistance

Alliance & Application Partner Program (AAPP)

Complete e-business solutions at your disposal

The Alliance & 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 Alliance & 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, who are labeled Alcatel-Lucent Compliant Application, come from every area of voice and data communications.

The Alliance & 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 Alliance & Application Partner, you can access the AAPP website at this URL:
http://www.applicationpartner.alcatel-lucent.com

Alcatel-Lucent.com

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

Alcatel-Lucent documentation

Alliance & Application Partner Program (AAPP)

If registered Alliance & Application Partner, you can access the current AAPP documentation at this URL:
http://www.applicationpartner.alcatel-lucent.com and then click the *Partner Center* link.

Alcatel-Lucent Business Partner Program (ABPP)

The Alcatel-Lucent Business Partner Program is designed to empower and maximize the business of the Partners. In addition, it enables them to help their customers successfully maximize their telecom investment through optimum deployment and proper configuration of Alcatel-Lucent's solutions. Alcatel-Lucent Partners also receive the added benefit of rapid, highly qualified service and support as well as world class training. Alcatel-Lucent will work closely with Business Partners to provide top quality design, delivery, and support of the very best solutions for your customers. The Business Partner Program is designed around a flexible and scalable framework so each Partner can identify the exact support they need. So, depending on your specific requirements you can quickly become a 'Certified', 'Expert' or 'Premium' Business Partner with one of the world leaders in the communications industry.

If registered Alcatel-Lucent Business Partners, you can access to an exciting on-line resource centre with a wealth of information on all product lines at this URL:
http://www.businesspartner.Alcatel-Lucent.com

Technical assistance

In order to guide you in your purchasing decisions and provide you with assistance for updating our Communication Server and Networking Infrastructure products and for commercial development, Alcatel-Lucent has created the **SUPPORT CENTER**. The **SUPPORT CENTER** is responsible for the management and routing of all your requests. It includes **e-Support** and a **Contact Centre** reserved for registered Alliance & Application Partner and Alcatel-Lucent Business Partners.

The **Contact Centre** is open 24 hours a day; 7 days a week and is available in 5 languages. This Call Centre has a team of 15 people and handles 10; 000 requests per month.

- e-Support from the Alliance & Application Partner Web site (if registered Alliance & Application Partner): http://www.applicationpartner.alcatel-lucent.com click the *Partner Center* link and then *Support*
- e-Support from the Alcatel-Lucent Business Partners Web site (if registered Alcatel-Lucent Business Partners): http://www.businesspartner.Alcatel-Lucent.com click the *e-Support* link and then *e-Service Request*
- e-mail: Support.Center@Alcatel-Lucent.fr
- Fax number: +33 (0) 3 90 67 73 45
- Telephone numbers:

Alcatel-Lucent Business Partner Contact Center:

France :	0 811 900 110	French agent
Austria :	0 810 810 012	German agent
Denmark :	70 11 21 09	English agent
Germany :	0 1 803 000 680	German agent
Ireland :	1 890 925 039	English agent
Italy :	848 800 389	Italian agent
UK :	0 845 601 4101	English agent
Spain :	901 120 085	Spanish agent
Switzerland :	0 844 850 588	German agent

For other countries:

English answer :	+ 33 (0)3 88 55 69 04
French answer :	+ 33 (0)3 88 55 69 02

Alcatel-Lucent training

Technological innovative cycles are quickening and your customers are more and more demanding regarding the quality of services. In order to meet these requirements, you have to invest in skills: a key success factor for services.

If registered Alcatel-Lucent Business Partners, you can access to the training part at this URL: http://www.businesspartner.Alcatel-Lucent.com and then click the *Training* link.

Our vision of learning services is described in the **Services Portfolio section**. The **Certification section** gives you some statistics and details on how training curricula are designed to match certification levels.
All updated training curricula and assessment tools are available in the **Curricula & Catalogues section**.
The **Schedule section** is regularly updated to show forthcoming training sessions over the world. The **How to Enroll section** provides you with the registration procedure and the Alcatel-Lucent University Customer Service list of contacts world wide.
Last but not least, find statistics and reports of what you think about our training services in the **Customer Satisfaction section**.

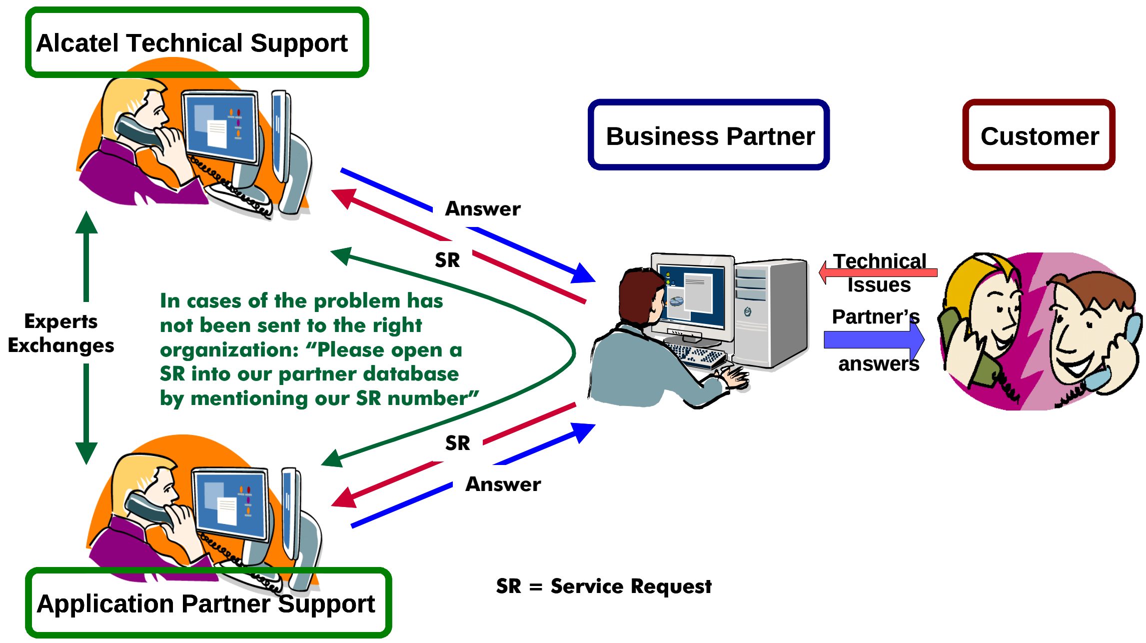
The Alcatel-Lucent commitment : enabling you to optimize your training investments.

Appendix E: Alcatel-Lucent escalation process in case of problem with a certified external application (referenced in the AAPP)

Introduction

The purpose of this document is to define the split of responsibilities and the escalation process to be applied by the Business Partners when facing a problem with a solution involving an Alcatel-Lucent platform and an external application ***with a valid Alcatel-Lucent Inter-working report.***

As for other Alcatel-Lucent equipment, the Alcatel-Lucent business partner is the only one facing the end-customer for support or maintenance. The Business partner will open cases (service request) either on Alcatel-Lucent side or on Application Partner side depending on the nature of the issue. Expert from both companies will collaborate to provide the best and quickest correction.



General Rules

The following general rules are applied:

- **Only certified AAPP applications are officially supported by Alcatel-Lucent**
- **The certification is based on tests suite passed by Alcatel-Lucent and the Application Partner and the result is consigned into an Inter-Working Report (IWR) validated by the two parties.**
- The IWR is available on the AAPP Web site.
- Only the major releases of both parties are certified. Certification tests are usually not performed for intermediate versions. Only the existence of the IWR in the AAPP Web site **for the right Alcatel-Lucent release** is the guarantee that the application has been certified with this Alcatel-Lucent release.
- If the IWR for the Alcatel-Lucent release is not available, Alcatel-Lucent doesn't engage any responsibility. In that case, please contact the central Pre-Sales team.
- The existence of the IWR engages Alcatel-Lucent **and the Application Partner**. Both parties are engaged, not exclusively Alcatel-Lucent (see the section escalation process).

Warning:
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 AAPP web site.

The escalation process

As stated above, the Alcatel-Lucent support will be limited to applications with a valid inter-working report. Known problems or remarks mentioned in the IWR will not be taken into account.
In case of problem, the two 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 Business Partner to the Alcatel-Lucent Hot-line via the standard process: open a ticket (Service Request –SR)

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

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

1) **The Application Partner shall be contacted first by the Business Partner** or the party responsible for that Application for an analysis of the problem.
Alcatel-Lucent has to be involved solely if the application partner demonstrate, with traces, after reproduction of the problem, that the defect which has generated the end-user's demand of support is coming from the equipment provided by Alcatel-Lucent or if he needs support of Alcatel-Lucent.

2) The Business partner will escalate the problem to the Alcatel-Lucent Hot-line if the Application Partner has demonstrated a problem on Alcatel-Lucent side or if the Application Partner (not the Business Partner) needs the involvement of Alcatel-Lucent.

In that case**, the Business Partner must provide the reference of the Case Number on 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 Application Partner side are insufficient or do no exist.

Note:

Involvement of the Business Partner is mandatory because the access to the Alcatel-Lucent Platform (remote access, login/password) is under the Business Partner responsibility.