ALCATEL OXE R 11.1 configuration guide for SIP trunks

configuration requirements

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Version 1.3

Document control

Date	version no.		change/addition
2014-07-22	1.0	L.Tomaszewski	Creation of the configuration guideline R10.0 based on [1]
2014-09-30	1.1	L.Tomaszewski	Appendix added (8082NOE, OT4135, 4059EE configuration)
2014-10-24	1.2	L.Tomaszewski	Customized configuration for the France market
2015-04-20	1.3	L.Tomaszewski	Document upgrade to OTBE R2.1/OXE 11.1

Table of contents

1 General

1.1 Scope of the document

The aim of this document is to provide configuration guideline for Alcatel OmniPCX Enterprise VoIP Support of IP Telephony BIV SIP series 2 service.

BIV SIP service concentrates on validating interoperability of IMS NSN infrastructure with Alcatel OmniPCX Enterprise solution.

The document presents configuration requirements on the component of IP Telephony Architecture in order to ensure the interoperability of Alcatel OmniPCX Enterprise solutions IP PBX with IMS NSN infrastructure.

1.2 References

Title/Web link	Location
[1] VISIT SIP Configuration Guideline for OXE	UCC ALU sharepoint
[2] Alcatel-Lucent OXE R10 SIP documentation	Alcatel Business Portal/ Technical Knowledge Base
[3] TC1865 ed.02 8082 MY IC PHONE FIRMWARE UPDATE PROCEDURE TO BE READY FOR OT CONNECTION FOR 8082 USAGE IF FIRMWARE LOWER THAN R260_01_012_6	Alcatel Business Portal
[4] TG0071-Ed02 / 8AL 90610 USABed02 4059EE Trouble shooting guide	Alcatel Business Portal

2 Configuration requirements

2.1 General

This OXE configuration guideline allows correct interoperability with the IMS NSN infrastructure.

This recommendation includes the following components:

- Media parameters
- Call Allowance control
- Call Routing to P-CSCF

2.2 Configuration principles

This chapter presents the configuration principles, the chapter 3 details all the screenshots of OXE.

2.2.1 Media parameters

To comply with the existing rules regarding the voice packetization on the IP network, the following components have to be configured:

- Type of codec: **G711** or **G729**
- Voice Payload: payload (number of bytes) included in each IP packet **160 bytes** (20ms) for G711 or **20 bytes** (20ms) for G729
- Silence Suppression: VAD disabled for both G711 and G729

Example with G711 codec / payload 160 bytes: each 20ms an IP packet with 160 bytes of voice is transmitted.

IP (20 bytes)	UDP (8 bytes)	RTP 12 bytes	Payload 160 bytes

DTMF are configured in RFC2833. All media parameters are accessible through the administration tools **OmniVista 8770** or **mgr**.

2.2.1.1 Fast Start

The Fast Start mode has to be enabled through the menu **IP > IP Parameters > Fast Start**. The parameter **Fast Start** has to be set to **true**. Boards (GD and/or INTIP) must be rebooted after configuration.

2.2.1.2 Payload 20ms

The payload used for codec G711 has to be configured through the menu **IP > IP Parameters > G711 VOIP framing**. The parameter **G711 VOIP framing** has to be set to **20ms**.

The payload used for codec G729 has to be configured through the menu **IP > IP Parameters > G729 VOIP framing**. The parameter **G729 VOIP framing** has to be set to **20ms**.

2.2.1.3 Round trip delay request

In default configuration, OXE sends a round trip delay request but some gateways don't understand this message. Due to identified problems, the round trip delay request has to be configured though the menu **IP > IP Parameters > Round trip delay request.** The parameter **Round trip delay request** has to be set to **false**. OXE boards (GD and/or INTIP) must be rebooted.

2.2.1.4 IP Domain configuration

Our recommendation for IP domain are the following (same as Alcatel-Lucent recommendation):

- IP Domain 0: the CS, CPU, AS are located by default here.

2.2.1.5 Compression Parameters

Quantification Law

The recommendation for quantification law is to use A law because we are in Europe.

On OXE, the law is defined in:

- Media Gateway
- System

In Media Gateway, the law has to be configured through the menu **Media Gateway > "select an media Gateway instance"**. The parameter **Law** has to be set to **Default** (Default = Law of the System).

In System, the law has to be configured through the menu **System > Other System Param. > D/H> System Parameters > Law**. The parameter **Law** has to set to **A Law.**

Voice Activity Detection

The Voice Activity Detection (VAD) has to be configured through the menu **System > Other System Param. > D/H> Compression Parameters**. Parameters **Voice Activity Detect (Comp Bds)** and **Voice Activity Detection on G.711** have to be set to false**.**

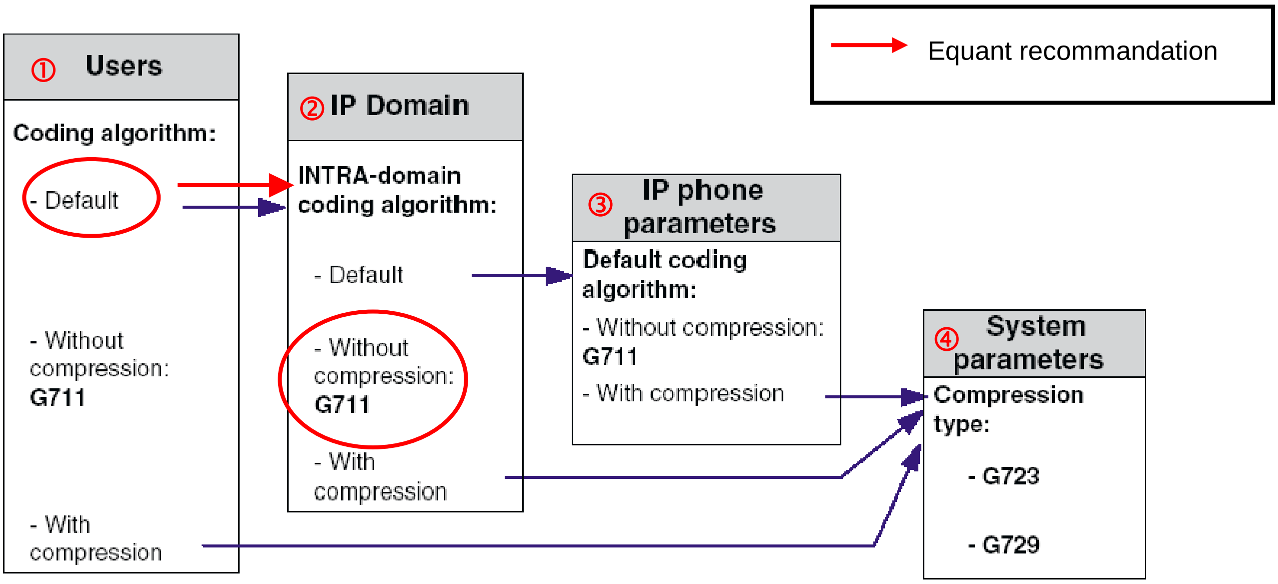
Codec Usage

The recommendation for codec usage is to use only G711 on the LAN and G711, G729 on the WAN.

In order, on OXE, the codec usage is defined by:

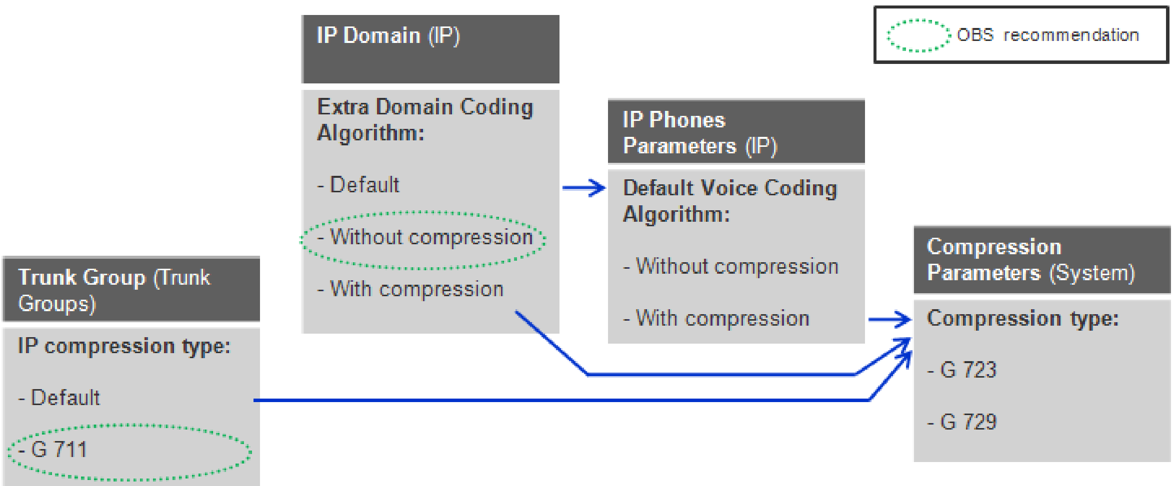
- Codec preference used by Trunk Group
- Codec preference used by Users
- Codec preference used by IP Domain
- Codec preference used by IP phone parameters
- Type of compression used by the system (G729/G723)
- Codec preference used by VPN Overflow

In case of intra-domain calls, the algorithm applied for the IP-Phone is determined as indicated in the figure below.

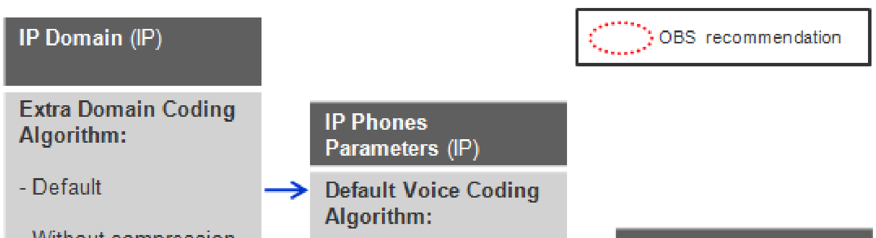


Compression for an Intra-lan call (intra-domain call)

In case of extra-domain calls, IP-Phone settings are not involved. Only trunk and domain settings apply, as indicated in figures below.



Compression for an Inter-lan call g711 (extra-domain call)



Compression for an Inter-lan call g729 (extra-domain call)

OBS recommendation for codec configuration is to use public SIP Trunk configuration in first step then IP Domain configuration. It is determined in the figure below.

The parameter for codec used by SIP trunk is accessible through the menu **Trunk Groups > Trunk Group > "select the SIP Trunk Group ID"**. The parameter **IP Compression Type** has to be set to **G711** (if G711 used) or **Default** (if G729 used)**.**

The parameter "Type of compression used by the system" is accessible through the menu **System > Other System Param. > D/H> Compression Parameters > Compression Type**. The parameter **Compression Type** has to be set to **G729**.

The parameter for codec preference used by users is accessible through the menu **Users > TSC IP Users > "select a user"**. The parameter **Voice Coding Algorithm** has to be set to **Default**.

Parameters about IP domain are accessible through the menu **IP > IP Domain**. First of all you have to define codec used for intra-domain and extra-domain calls. Parameters **Intra-domain Coding Algorithm** has to be set to **Without Compression** (G711) and **Extra-domain Coding Algorithm** has to be set to **Without Compression** (if G711 used) or **With Compression** (if G729 used). After that, you can define your IP Domain through the menu **IP > IP Domain > IP Domain Address.** An IP Domain is defined by an IP range. You can allocate devices (IP Phones, CPU, boards …) to an IP Domain if you include IP address devices in the IP range of IP Domain. To create the IP range in an IP Domain, put IP address in parameters **IP Address Low** and **IP Address High** (Format X.X.X.X).

>> IP Domain 0 is the default IP Domain, Call Server IP Address must be declared on the Domain 0, the same if an external voicemail (4645/8440) is used

The parameter for codec preference used by VPN Overflow is accessible through the menu **Inter-Node links > VPN Overflow**. **.** The parameter **IP Compression Type** has to be set to **G711** (if G711 used) or **Default** (if G729 used)**.**

2.2.1.6 Calling Name / Display presentation / Callback

Country code has to be configured through the menu **System > Other System Param > D/H > Signaling String** in order to distinguish national and international calls. To see the calling name presentation on the set display of phones for incoming calls (**national**), the parameter "**DEF**" has to be added through the menu **Translator > External Numbering Plan > Ext. Callback Translation**.

To see the calling name presentation on the set display of phones for incoming calls (**intenational**), the parameter "**A**" has to be added through the menu **Translator > External Numbering Plan > Ext. Callback Translation**.
To see only the display presentation on the set display of phones for incoming calls, parameters "DEF" and "A" have to be deleted through the menu **Translator > External Numbering Plan > Ext. Callback Translation**.Calling name sending over SIP Trunk toward IMS is not recommended in BIV SIP service and it should be disabled. Parameter to block sending display name to external calls is accessible through the menu **System > Other System Param > External Signaling Parameters**. The parameter **Calling Name Presentation** has to be set to false..

>> This parameter is not used in SIP-Trunking

Note 1:

It is possible to remove or add digits following the type of the received number:

- DEF : Default type;
- A : International type;
- B : Private type;
- <number>.

Note 2:

To use callback prefix, be careful with management of Ext. Callback Translation.

2.2.1.7 Display for forward scenarios

To see the calling name presentation on the set display of called phone in forward scenarios, the parameter **NPD for external forward** has to be set to a value **different from -1** through the menu **System > Other System Param. > D/H > External Signaling Parameters > NPD for external forward**.

2.2.1.8 Caller secret identity

As soon as a user would like dial any number, caller secret identity can be became enabled "on demand" of the user, or automatically.

Regarding the method "on demand" of the user:

Accessible through the menu **Classes of Service > Phone Features COS > 0** (corresponding to the class of service of the device)
Parameter **"Secret/Identity"** has to be set to **1** in order to enable secret identity of caller on demand of this one.

Regarding the automatic method:

Accessible through the menu **Entities > then select the correct entity** which match with the correct device
Parameter **"Caller ID secret"** has to be set to **Yes** in order to enable automatically secret identity of all devices linked of this entity.

2.2.1.9 Timer 42

Accessible through the menu **System > Timers > 42**

The **timer 42** has to be set to value **5**. This timer permits to speed up the display on the set (in order to send DTMF earlier or perform a second call, etc.)

2.2.1.10 UDP lost

When the CS is cut from the network, after 32s the CS is up again. If the timer "UDP_Lost + UDP_Lost_Reinit" is lower than 32s, the boards will reboot.

The implementation of **UDP_Lost** timer isn't mandatory. We recommend to put to **45s** :

- a Quality of Service COS has to be set up for each GD/GA/INTIPA :

  Shelf > Board > Ethernet Parameters > "select an INTIP or GD/GA board" > IP Quality of Service = **0**

- a Quality of Service COS has to be set up a for all boards and IPPhone inside their own IP Domain :

  IP > IP Domain > "select an IP Domain" > IP Quality of Service = **0**

- Finally, UDP lost timer has to be set up for each Quality of Service COS :

  IP > IP Quality of Service COS > "select CoS QoS number 0" > UDP Lost = 45s

2.2.2 Call Routing

2.2.2.1 VoIP Numbering plan

It is required to use a public dial plan (as described below) for BIV SIP.

Outgoing Call (IPBX =>IMS)	Incoming Call (IMS=> IPBX)
From=+33ZABPQMCDU** RURI/To = same format as PSTN	From=+33ZABPQMCDU or +CCNSN RURI/To=+33ZABPQMCDU

- private number (if all PN have associated SDA and private numbering plan is collected)
- +CCNSN
- +33ZABPQMCDU or 0033ZABPQMCDU even if calling party is in France

(**) From must be either DID (NDS) number of calling phone or default public number (NDI) of calling site if calling phone doesn't have DID number

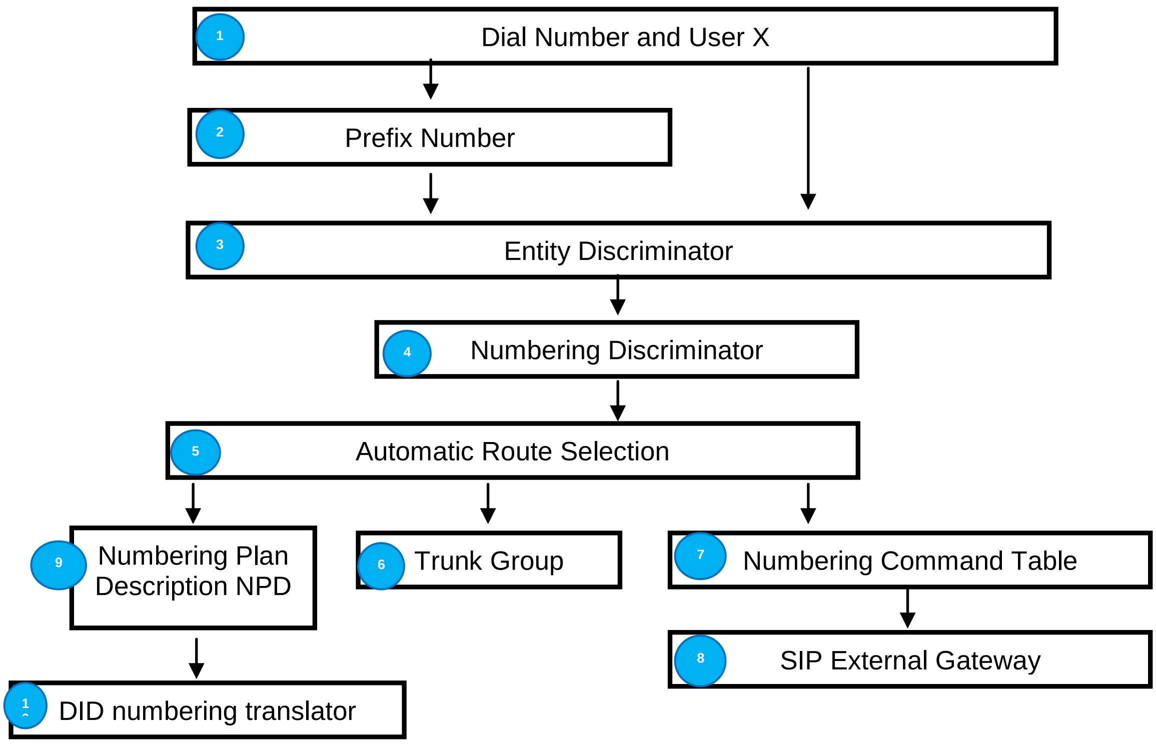
2.2.2.2 Route mechanisms on OXE

Call routing decides what path an outgoing call takes using the digits that are dialed. This Configuration Guideline focuses only on call routing mechanisms for On-net calls to Off-net calls.

Call Routing mechanisms for On-net to Off-net calls

For Outgoing calls, public SIP Trunk configured in ARS Route is used.

There are 10 steps for On-net calls to Off-net calls. The following chart shows the logic of the configuration of SIP trunk configuration. Steps 1 to 10 describe the call routing mechanisms for public numbering plan. If it's chosen to use a private numbering plan, follow only steps 1 to 8.



Below, are mentioned the explanations concerning the configuration of each object.

1 Dial Number:
When a user dials a number, this number matches a **prefix** and the user matches an **Entity.**2 Prefix Plan:
The prefix plan has to be configured through the menu **Translator > Prefix Plan > Create > "Prefix instance number", type ARS Prof Trg Grp Seiz with overlap**. The parameter **Discriminator No.** has to match a discriminator number in one Entity.

>> Add a screenshot

Moreover, a PCX prefix must be configured on DPNSS mode through following menu **Translator > Prefix Plan > "Prefix instance number"**.

3 Discriminator Selector:
When a user dials a number, the user matches an **Entity Number** in **User** parameters. The Entity Discriminator has to be configured through the menu **Entities > D/H > Discriminator Selector > "select an Entity Number"**. The Discriminator No. has to match a **Numbering Discriminator**.

4 Discriminator Rule:
The numbering discriminator has to be configured through the menu **Translator > External Numbering Plan > Numbering Discriminator > D/H > Discriminator Rule > "create a Discriminator No."**. The parameter **ARS Route List Number** has to match the **ARS Route**. Number of digits different than -1

4-bis

Go to translator > Automatic Route Selection > Numbering Command Table > Create

```
┌Review/Modify: Numbering Command Table──────────────────────────────────────┐
│                                                                             │
│             Node Number (reserved) : 1                                      │
│                Instance (reserved) : 1                                      │
│                Instance (reserved) : 1                                      │
│                           Table ID : 1                                      │
│                                                                             │
│                  Carrier Reference : 1                                      │
│                            Command : ---------------------------------------- │
│         Associated Ext SIP gateway : 1                                      │
│                                                                             │
```

5 ARS Route :
The ARS matched by **Discriminator number** has to be configured through the menu **Translator > Automatic Route Selection > ARS Route list > D/H > ARS Route > "create a ARS Route list"**. The parameter Trunk Group has to match the public SIP Trunk Group and the parameter Numbering Command Tabl. ID has to match a SIP external gateway (4-bis). ARS has to be set to Speech and Fax mode.6 Trunk Groups:
The SIP Trunk Group has to be configured through the menu **Trunk Groups**. The parameter **Trunk Group Type** has to be set to **T2, Q931 Signal variant: ISDN All Countries** and **T2 Specification** has to be set to **SIP**.

Note: Remote Network number selected must not be used on another Trunk Group

The number of access associated to this trunk has to be configured through the menu **Trunk Groups > Trunk Group > Virtual accesses for SIP > "select a Trunk Group ID"**.

see 4-bis8 SIP Ext Gateway:
Accessible through the menu **SIP > SIP Ext Gateway**.
The parameter **SIP Remote domain, Belonging Domain** and **Outgoing Realm** have to match the **domain name** of the IMS network.
Number of the port **5060** to which the OXE sends SIP messages toward P-CSCF has to be configured through the parameter **SIP Port Number**.
The parameter **SIP Transport Type** has to be set to **UDP** as required by the IMS NSN infrastructure.
Registration ID has to match IPBX NDI trunk number in format +33ZABPQMCDU and **Registration timer** equal to 3600.

SIP Outbound Proxy has to be set up to FQDN of the P-CSCF which is resolved by DNS servers which IP addresses have to be configured in **SIP DNS1 IP Address** and **SIP DNS2 IP Address** fields. **DNS type** has to be configured as **DNS A.**
Outgoing username and **Outgoing Password** are used to authorize outgoing SIP requests.
Outgoing username has to match the form **+33ZABPQMCDU@Domain_Name**, where +33ZABPQMCDU is NDI number of the SIP trunk and Outgoing Password must match the one provided by a network administrator.

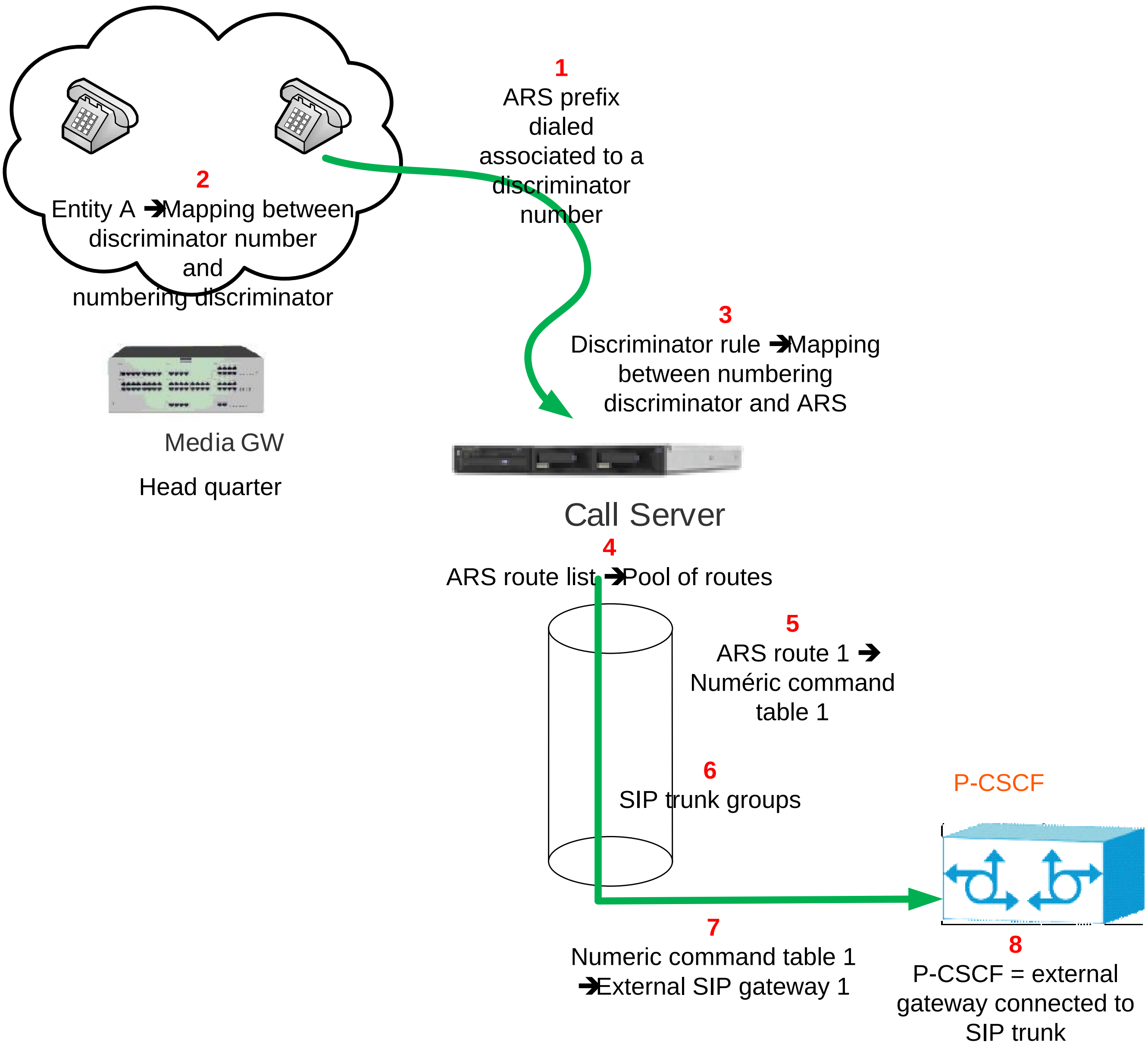
The parameter **RFC 3325 supported by the distant** has to be set to **, why False?** To be compliant with SIP profile, SDP mustn't be present in 180 ringing sent by OXE. The parameter **SDP in 18x** has to be set to **false**. The parameter **Minimal authentication method** has to be set to **SIP None**. To be compliant with SIP profile, the parameter **Dynamic Payload type for DTMF** has to be set to **101**.

The provisional acknowledge responses (PRACK) separated in two new others **100REL for outbounds calls** and **100REL for incoming calls**. As the IMS does not support PRACK, both parameter have to be set up to **Not Supported**.

No probing is required by the IMS as this function is handled by registration process, then the parameter **Supervision timer** has to be set to **0**.

The parameter **Send only trunk group algo** has to be set to **false** in order to send both G711 and G729 codec offer.

The parameter **Trunk Group number** has to match the **public SIP Trunk Group** in order to route incoming calls from the IMS.
The parameter Support Re-invite without SDP has to be set to TRUE in order to optimize capabilities exchanges in case of transfer offnet – offnet**9** Numbering Plan Description:
Accessible through the menu **Translator > External Numbering Plan > Numbering Plan Description**. > Create
The numbering plan description translates the private call numbers and builds the outgoing call numbers through these parameters: Calling/Called Numbering Plan Identifier and Calling/Called DID Identifier.
Calling Numbering plan ident. ?
Called numbering plan ident. +
10 DID numbering translator:
Accessible through the menu **Translator > External Numbering Plan > DID Numbering Translator**.
The DID numbering translator matches the internal number to the external number,



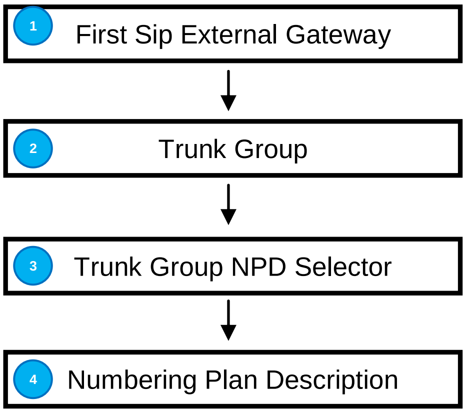
General principle

Call Routing mechanisms for Off-net to On-net calls

For Incoming calls, public SIP Trunk configured in SIP External Gateway is used.

Incoming Calls Routing Mechanisms

There are 4 steps for Off-net calls. The following chart shows the logic of the configuration of SIP trunk configuration.



3 Trunk Group NPD Selector:
Accessible through the menu **Trunk Groups > D/H > Trunk Group NPD Selector"**. The parameter associates the trunk group to a public NPD id.4 Numbering Plan Description:
Accessible through the menu **Translator > External Numbering Plan > Numbering Plan Description**.
The numbering plan description translates the incoming call numbers and builds the internal call numbers through these parameters: Calling/Called Numbering Plan Identifier and Calling/Called DID Identifier.
The same as previsously?

2.2.2.3 Calling number presentation for Forwarded Calls

It is recommended for each new installation or migration to activate the diversion field for external forward. To perform, three parameters have to be activated:

To configure initial caller number to be forwarded by forwarding party the parameter "NPD for external Forward" set up to value different than "-1", accessible through the menu **System > Other System Param > External system parameters > NPD for external forward**

Note: **Diversion** and **History-Info** headers have to be disabled by IPBX. For that the parameter "IE External Forward", accessible through the menu **Trunk Groups > Trunk Group** must be setup to value None.

2.2.3 4645 Voice Mail

The payload used for 4645 (codec G711) has to be configured through the menu **IP > IP Parameters > G711 VOIP Framing for 4645**. The parameter **G711 VOIP Framing for 4645** has to be set to **20ms** (**only supported for Appliance Servers and CS2 boards**).

Note (restriction): For CS1 (Common Hardware first release), **"G711 VOIP Framing for 4645"** parameter has to remain in the default configuration (**30ms**).

2.2.4 4059IP integration

2.2.4.1 RBT tone

The parameter **Tone presence** has to be activated through the menu **Attendant > Attendant sets** in order to have a ringing tone in the handset.

2.2.4.2 Welcome guide

To have a welcome message, the **timer 102** has to be modified (different to value **0**) through the menu **System > Timers > 102**.

3 Configuration screens

Note: The **static parameters** are in blue and in bold while the *variable parameters* are in blue and in italic in this chapter.

3.1 Media Parameters

3.1.1 Fast Start

Accessible through the menu **IP > IP Parameters > Fast Start**.

IP Parameters

…

System Option + Fast Start

Fast Start + True

parameter has to be set to ***True***

3.1.2 Payload 20ms

To comply with our requirements:

- Payload for G.711 has to be set to 20ms through the menu **IP > IP Parameters > G711 VOIP Framing**.

IP Parameters

…

System Option + G711 VOIP Framing

G711 VOIP Framing + 20 ms

G711 payload has to be set to ***20ms***

- Payload for G.729 has to be set to 20ms through the menu **IP > IP Parameters > G729 VOIP Framing**.

IP Parameters

…

System Option + G729 VOIP Framing

G729 VOIP Framing + 20 ms

G729 payload has to be set to ***20ms***

3.1.3 Round trip delay request

To comply with our requirements, the Round trip delay request has to be disabled because some gateways don't understand this message. This parameter is accessible through the menu **IP > IP Parameters > Round trip delay request**.

IP Parameters
…
System Option + Round trip delay request

Round trip delay request + False

*parameter has to be set to **False***

3.1.4 Compression Parameters

3.1.4.1 Quantification Law

Media Gateway:

Accessible through the menu **Media Gateway > "select an media Gateway instance"**.

Media Gateway
…
Main shelf type + Media Gateway Large
Law + Default
Reference + YES
First expansion shelf type + None
Second expansion shelf type + None

*parameter has to be set to **Default***

System:

Accessible through the menu **System > Other System Param. > System Parameters > Law**.

System Parameters
…
System Option + Law

Law + A Law

*parameter has to be set to **A Law***

3.1.4.2 VAD

To comply with our requirements, the Voice Activity Detection has to be disabled through two parameters:

1. First in **System > Other System Param. > Compression Parameters > Voice Activity Detect (Comp Bds)**.

Compression Parameters
…
System Option + Voice Activity Detect (Comp Bds)

Voice Activity Detect (Comp Bds) + False

*parameter has to be set to **False***

2. Second in **System > Other System Param. > Compression Parameters > Voice Activity Detection on G711**.

```
┌─ Compression Parameters───────────────────────────────┐
|…
|System Option + Voice Activity Detection on G711
|
|Voice Activity Detection on G711 + False
└────────────────────────────────────────────────────────┘
```

*parameter has to be set to **False***

3.1.4.3 Codec Usage

To comply with the existing rules regarding the voice packetization on the IP VPN network, the following components have to be configured:

- Type of codec: **G711/ G729**
- **Intra LAN** calls in **G711**
- **Inter LAN** calls in **G711/ G729**

Compression Type:

Accessible through the menu **System > Other System Param. > Compression Parameters > Compression Type**.

```
┌─ Compression Parameters───────────────────────────────┐
|…
|System Option + Compression Type
|
|Compression Type + G 729
└────────────────────────────────────────────────────────┘
```

*parameter has to be set to **G729***

Codec by Users:

Accessible through the menu **Users > TSC IP Users > "select an user"**.

```
┌─ TSC IP User──────────────────────────────────────────┐
|…
|Directory Number : 1521101
|Directory Number : 1521101
|
|Set Type + IPTouch 4068
|Voice Coding Algorithm + Default
|Terminal Ethernet Address : 00:80:9f:78:20:ac
|IP Address : 10.152.101.21
|IP Domain Number : 0
|Use of volume in system + YES
|Reset For Update Authorized + YES
|IP-Softphone Emulation + NO
└────────────────────────────────────────────────────────┘
```

*parameter has to be set to **Default***

Codec by IP Domain:

Accessible through the menu **IP > IP Domain > "select an IP Domain"**.

- G711 configuration:

```
┌─ IP domain────────────────────────────────────────────┐
|…
```

*Intra-Domain Coding Algorithm has to be set to **Without Compression***

```
IP Domain Number : 0

IP Domain Name : -------------------
Country + Default
Intra-domain Coding Algorithm + Without Compression
Extra-domain Coding Algorithm + Without Compression
FAX/MODEM Intra domain call transp + NO
FAX/MODEM Extra domain call transp + NO
Domain Max Voice Connection : -1
IP Quality of service : 0
Contact Number : -----------------------------
Backup IP address : ---------------------------------------------
Trunk Group ID : -1
IP recording quality of service : 0
Time Zone Name + System Default
Calling Identifier : -----------------------------
Supplement. Calling Identifier : -----------------------------
SIP Survivability Mode + NO
```

*Extra-Domain Coding Algorithm has to be set to **Without Compression***

- G729 configuration:

```
IP domain
…
IP Domain Number : 0

IP Domain Name : -------------------
Country + Default
Intra-domain Coding Algorithm + Without Compression
Extra-domain Coding Algorithm + With Compression
FAX/MODEM Intra domain call transp + NO
FAX/MODEM Extra domain call transp + NO
Domain Max Voice Connection : -1
IP Quality of service : 0
Contact Number : -----------------------------
Backup IP address : ---------------------------------------------
Trunk Group ID : -1
IP recording quality of service : 0
Time Zone Name + System Default
Calling Identifier : -----------------------------
Supplement. Calling Identifier : -----------------------------
SIP Survivability Mode + NO
```

*Intra-Domain Coding Algorithm has to be set to **Without Compression***

*Extra-Domain Coding Algorithm has to be set to **With Compression***

Accessible through the menu **IP > IP Domain > IP Domain Address**. This menu allows defining a range of IP addresses belonging to an IP domain.

Call server must necessarily belong to domain 0.

```
IP Domain Address

Node Number (reserved) : 202
Instance (reserved) : 1
IP Domain Number : 0
IP Address Low : 6.4.36.1
```

IP Address High : 6.4.36.254
IP NetMask : *255.255.255.0*
IP Address Type + IP Range

Codec by Trunk Group:

Accessible through the menu **Trunk Groups > Trunk Group**:

- G711 configuration:

Trunk Group
…
Trunk Group ID : 106
Instance (reserved) : 1

Trunk Group Type + T2
T2 Specification + SIP
Public Network Ref. : ------
VG for non-existent No. + YES
Entity Number : 100
Supervised by Routing + NO
VPN Cost Limit for Incom.Calls : 0
Immediate Trk Listening if VPNCall + YES
VPN TS % : 50
CSTA-Monitored + NO
Max.% of trunks out CCD : 0
Ratio analog.to ISDN cost : ------
TS Distribution on Accesses + YES
Quality profile for voice over IP + Profile #1
IP Compression Type + G 711
IE for external forward + Nothing

public SIP Trunk (106 in the example)

*parameter has to be set to **G711***

- G729 configuration:

Trunk Group
…
Trunk Group ID : 106
Instance (reserved) : 1

Trunk Group Type + T2
T2 Specification + SIP
Public Network Ref. : ------
VG for non-existent No. + YES
Entity Number : 100
Supervised by Routing + NO
VPN Cost Limit for Incom.Calls : 0
Immediate Trk Listening if VPNCall + YES
VPN TS % : 50
CSTA-Monitored + NO
Max.% of trunks out CCD : 0
Ratio analog.to ISDN cost : ------
TS Distribution on Accesses + YES
Quality profile for voice over IP + Profile #1

public SIP Trunk (106 in the example)

*parameter has to be set to **Default***

| **IP Compression Type + Default**
| IE for external forward + Nothing

3.1.5 Calling Name / Display presentation / Callback

Country code has to be configured through the menu **System > Other System Param > Signaling String** in order to distinguish national and international calls.

Signaling String
| …
| System Option String + SG Country Code
|
| System Option String Max Length : 8
| **Country Code :** *33*

Country Code (33 in the example if site in France)

The parameter "DEF" has to be added through the menu **Translator > External Numbering Plan > Ext. Callback Translation** in order to see the calling name presentation on the set display of phones for incoming calls (**national**).

Ext.Callback Translation
| …
| **Basic Number : DEF**
|
| **No.Digits To Be Removed :** *0*
| **Digits To Add :** *00*

digits to be removed according to incoming calls (0 in the example)

digits to be added according to incoming calls (00 in the example)

The parameter "A" has to be added through the menu **Translator > External Numbering Plan > Ext. Callback Translation** in order to see the calling name presentation on the set display of phones for incoming calls (**international**).

Ext.Callback Translation
| …
| **Basic Number : A**
|
| **No.Digits To Be Removed : 1**
| **Digits To Add :** *000*

digits to be removed according to incoming calls (1 in the example)

digits to be added according to incoming calls (000 in the example)

Parameters "DEF" and "A" have to be deleted through the menu **Translator > External Numbering Plan > Ext. Callback Translation** if the user wants to see only the display presentation on the set display of phones for incoming calls.

Note 1:

To send display name to external calls, the parameter **Calling Name Presentation** has to be activated through the menu **System > Other System Param > External Signaling Parameters**.

External Signaling Parameters
| …
| System Option + Calling Name Presentation
|

*parameter has to be set to **True***

Calling Name Presentation + True

Note 2:

It is possible to remove or add digits following the type of the received number:

- DEF : Default type;
- A : International type;
- B : Private type;
- <number>.

Note 3:

Menu Ext. Callback Translation has to be managed to use callback prefix.

3.1.6 Display for forward scenarios

General case

Accessible through the menu **System > Other System Param. > External Signaling Parameters > NPD for external forward**.

External Signaling Parameters
…
System Option + NPD for external forward

NPD for external forward : *33*

*parameter has to be set to a value different from **-1** (33 in the example)*

3.1.7 Caller secret identity

As soon as a user would like dial any number, caller secret identity can be became enabled “on demand” of the user, or automatically.

“On demand” of the user:

Accessible through the menu Classes of Service > Phone Features COS > 0 (corresponding to the class of service of the device)

Phone Features COS
…
Secret/Identity: 1
…

parameter has to be set to 1 in order to enable secret identity of caller on demand of this one

Automatically:

Accessible through the menu Entities > then select the correct entity which match with the correct device

Entities

parameter has to be set to Yes in order to enable automatically secret identity of all devices linked of this entity.

```
│...
│Caller ID secret + Yes
│...
└
```

3.1.8 Timer 42

Accessible through the menu **System > Timers > 42**.

```
┌─ Timers ─┐
│...
│Timer No. : 42
│
│Timer units : 5
└
```

*parameter has to be set to **5***

3.1.9 UDP Lost timer

First, you have to set up a Quality of Service COS for each GD/GA/INTIP through the menu **Shelf > Board > Ethernet Parameters > "select an INTIP or GD/GA board"**.

```
┌─ Ethernet Parameters ─┐
│...
│Interface Type + GD
│Board IP Address : 6.4.36.5
│IP NetMask  : 255.255.255.0
│Default Gateway IP Address : 6.4.36.254
│IP Quality of service : 0
│Cryptographic box address : ---------------------------------------------------
│Board Ethernet Address : 00:80:9F:81:DB:12
│Interworking with Gatekeeper + YES
│Gatekeeper ID : 1
│Numb. of sig. channels IP Phones : 0
│Numb. of sig. channels inter-ACT : 0       -
│Board Ethernet Address : 00:80:9F:8
│IP Domain Number : 0
│E164 Number List Index : -1
│Gateway H323 name : --------------------------------
└
```

*parameter has to be equal to **0***

Then, you have to set up a Quality of Service COS for all GD/GA/INTIP and IP phones inside their own IP Domain through the menu **IP > IP Domain > "select an IP Domain"**.

```
┌─ IP domain ─┐
│...
│IP Domain Name : --------------------
│Country + Default
│Intra-domain Coding Algorithm + Without Compression
│Extra-domain Coding Algorithm + Without Compression
│FAX/MODEM Intra domain call transp + NO
│FAX/MODEM Extra domain call transp + NO
│Domain Max Voice Connection : -1
│IP Quality of service : 0
```

*all IPPhones and boards in this IP Domain will have the IP QoS Cos equal to **0***

Contact Number : -----------------------------
Backup IP address : ---
Trunk Group ID : -1
IP recording quality of service : 0
Time Zone Name + System Default
Calling Identifier : -----------------------------
Supplement. Calling Identifier : -----------------------------
SIP Survivability Mode + NO

Finally, you set up UDP lost timer for each Quality of Service COS through the menu **IP > IP Quality of Service COS > "select CoS QoS number 0"**.

IP Quality Of Service COS
…
IP QoS COS : 0

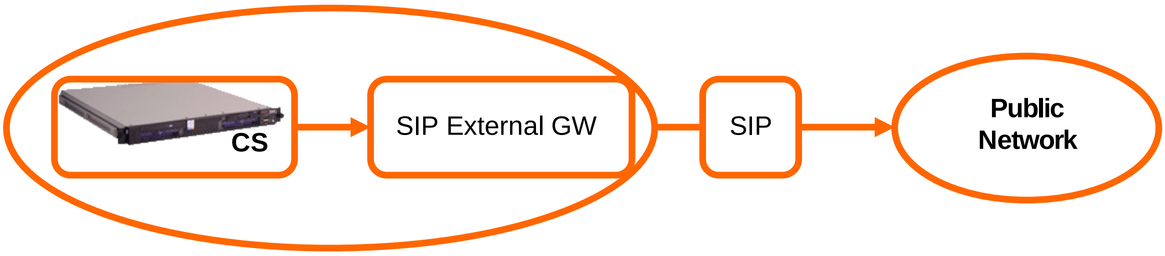
Quality of Service Category Name : --------------------
8021Q Used + False
8021p Priority : 5
VLAN ID : 0
TOS/diffServ : 46
UDP Lost : 45
UDP Lost Reinit : 7
UDP Keep-alive : 15
SIP Diff. Service : 40

*for the IP QoS CoS 0, the UDP Lost time is set to **45s***

3.2 Routing mechanisms on OXE

3.2.1 Call Routing mechanisms for On-net to Off-net calls

To understand call routing mechanisms with Global Gateway, we will follow a call example (call from IP Phone “1009” to “0223XXXXXX”).



There are 19 steps for On-net calls to Off-net calls:

Step 1 Dial Number:
When a user dials a number, this number matches a **prefix** and the user matches an **Entity**.

Step 2 SIP Gateway:
The **internal** SIP gateway is necessary to start SIP processes. There is only one gateway running on the CS. The SIP gateway has to be configured through the menu **SIP > SIP Gateway**.

Note: It is advised to declare **SIP Subnetwork** and **SIP Trunk Group** parameters, even if they are not necessary for BIV SIP (as they are only used for SIP endpoints : SIP Phones, Voice Mail 8440 for instance). SIP Trunk Group configurable through the menu **SIP > SIP Gateway** matches the **private SIP Trunk** used by SIP endpoints.

Session Timer is too low, Min Session Timer: 1800, Session Timer: 3600

SIP Gateway

…
SIP Subnetwork : *15 (/or -1)*
SIP Trunk Group : *1(/or -1)*
IP Address : 6.4.36.1
Machine name - Host : BIV-OXE
SIP Proxy Port Number : 5060
SIP Subscribe Min Duration : 600
SIP Subscribe Max Duration : 86400
Session Timer : 300
Min Session Timer : 300
Session Timer Method + UPDATE
DNS local domain name : --
DNS type + DNS A
SIP DNS1 IP Address : 80.12.10.156
SIP DNS2 IP Address : 80.12.10.152
SDP IN 180 + True
Cac SIP-SIP + False
INFO method for remote extension + False
Dynamic Payload type for DTMF : 101

see « Note » above

maximum amount of time before a session is considered terminated / UPDATE is sent before SIP Session Timer expiry.

minimum value of the session timer accepted by the gateway – value set to Session Time/2

methods provided by the SIP gateway as session refresh requests

*SDP **must** be present in 180 ALERTING sent by the OXE to the end point*

*parameter has to be set to **False.** CAC is managed by the network AS*

*dynamic payload proposed in RFC 2833 / recommended value is **101***

Step 3 SIP Proxy:
Accessible through the menu **SIP > SIP Proxy**.

Default value of minimal authentication method is digest, not SIP None

Default value of Only authenticated incoming calls is TRUE, not False

SIP Proxy

…
SIP initial time-out : 500
SIP timer T2 : 4000
Dns Timer overflow : 5000
Recursive search + False
Minimal authentication method + SIP None
Authentication realm : --------------------------------------
Only authenticated incoming calls + False
Framework Period : 3
Framework Nb Message By Period : 25
Framework Quarantine Period : 1800
TCP when long messages: False
Retransmission number for INVITE : 5

*parameter has to be set to **500** / attribute specifies the initial value in milliseconds of the request/reply SIP message retransmission timeout. The retransmission timeout doubles after each retransmission.*

*TCP when long messages parameter has to be set to **False**, in order to disable a possible change-over through TCP.*

Step 4 Trunk Groups:
Accessible through the menu **Trunk Groups**.

Trunk Groups

…
Trunk Group ID : *106*

Trunk Group Type + T2
Trunk Group Name : *BIV_SIP*
UTF-8 Trunk Group Name : --------------------------------------
Number Compatible With : -1
Remote Network : 0
Shared Trunk Group + False
Special Services + Nothing
Node number : *2*
Transcom Trunk Group + False
Auto.reserv.by Attendant + False
Overflow trunk group No. : -1
Tone on seizure + False
Private Trunk Group + False
Q931 Signal variant + ISDN all countries
SS7 Signal variant + No variant
Number Of Digits To Send : 0
Channel selection type + Quantified
Auto.DTMF dialing on outgoing call + NO
T2 Specification + SIP
Homogenous network for direct RTP + NO
Public Network COS : 31
DID transcoding + False
Can support UUS in SETUP + True

Implicit Priority

public SIP Trunk (106 in the example)

*parameter has to be set to **T2***

name of Trunk Group (BIV_SIP in the example)

parameter has to match a Remote Network (0 in the example)

node number (2 in the example)

*parameter has to be set to **ISDN all countries***

*parameter has to be set to **SIP***

Activation mode : 0
Priority Level : 0

Preempter + NO

ncoming calls Restriction COS : 10
utgoing calls Restriction COS : 10
Callee number mpt1343 + NO
Overlap dialing + YES
Call diversion in ISDN + NO

Step 5 Trunk Group:
Accessible through the menu **Trunk Groups > Trunk Group > "select a Trunk Group ID"**.

Trunk Group

…
Trunk Group ID : *106* — *public SIP Trunk (106 in the example)*
Instance (reserved) : 1

Trunk Group Type + T2 — *parameter has to be set to **T2***
T2 Specification + SIP — *parameter has to be set to **SIP***
Public Network Ref. : ------
VG for non-existent No. + YES
Entity Number : *100* — *entity of site (200 in the example)*
Supervised by Routing + NO
VPN Cost Limit for Incom.Calls : 0
Immediate Trk Listening if VPNCall + YES
VPN TS % : 50
CSTA-Monitored + NO
Max.% of trunks out CCD : 0
Ratio analog.to ISDN cost : ------
TS Distribution on Accesses + YES
Quality profile for voice over IP + Profile #1
IP Compression Type + G 711 — *parameter has to be set to **G711** if G711 used*
*(*) parameter has to be set to **Default** if G729 used*
Use of volume in system + YES
Announcement for dial tone + NO
Announcement for Ring tone + NO
Private to Public Overflow + YES
End-to-end dialing + NO
DTMF end-to-end signal. + NO
Trunk group used in DISA + NO
DISA Secret Code : ----
Routing To Manager + NO
Trunk COS : 31
Sending of Progress message + YES
No. of digits unused (ISDN) : 0
B Channel Choice + YES
Channels: Attendant Control (Rsvd) : 0
Redirection For ACD (Dissuasion) + NO
DTO joining + NO
Consultation Call On B Channel + NO
Automated Attendant + NO

```
| Calling party Rights COS : 0
| TS Overflow + YES
| Number To Be Added : --------
| Charge Calling And ADN Creation + YES
| Logical Channel + 1__15 & 17__31
| Use Split Access + NO
| Heterogeneous Remote Network + NO
| COS Restrictions - Barring mode  + Not Restricted / Not barred
| ARS Class of service : 31
| External Access Server + NO
| CSTA Tracking MCDU Trk : --------
| IE for external forward + None
```

*Diversion and History-info upon forward has to be disabled. The parameter has to be set up to "**NONE**"*

Step 6 Virtual accesses for SIP:
The number of SIP accesses associated to this trunk has to be configured through the menu **Trunk Groups > Trunk Group > Virtual accesses for SIP > "select a Trunk Group ID"**.

```
Virtual accesses for SIP
| …
| Trunk Group ID : 106
| Instance (reserved) : 1
| Instance (reserved) : 1
|
| Number of SIP Accesses : 2
```

public SIP Trunk (106 in the example)

parameter has to be define between 2 (=60 simultaneous calls) and 32 (=960 simultaneous calls)

Step 7: Numbering Plan Description.
Accessible through the menu **Translator > External Numbering Plan > Numbering Plan Description (NPD) > "select a NPD identifier"**.

Numbering Plan Description (NPD)
…
Description identifier : *33*

Name : *BIV_SIP _NPD*
Calling Numbering plan ident. + *NPI/TON ISDN National*
Called numbering plan ident. + *NPI/TON ISDN Unknown*
Authorize personal calling num use + False
Install. number source + Entity source
Default number source +Entity source
Called DID identifier : 1
Calling/Connected DID identifier : 1

NPD identifier (33 in the example)

Numbering Plan Description name BIV_SIP NPD in the example)

parameter has to be set to ISDN

parameter has to be set to Entity source

parameter has to be set to Entity source

Those parameters have to be set to 1.

Called DID identifier provides the match between internal and external numbers through a DID number translator rules.
In this case, the numbering format will be the following:

- Incoming call (from SBC to IPBX) :
  - RURI / To = **+33ZABPQMCDU**
  - PAI / From = **+33ZABPQMCDU** or **+CCNSN**
- Outgoing call (from IPBX to SBC) :
  - RURI / To = same format as PSTN network
  - PAI / From = **+33ZABPQMCDU**

Translator > External Numbering Plan > DID Numbering Translator

DID Numbering Translator
…
DID num. transl. identifier : 0
Name : LIMITED_LOCAL_DID
DID Translator Mode + Default DID mode

DID number translator Identifier

DID Number Translator rules
…
DID num. transl. identifier : 0
First External Number : 33296084675
First Internal Number : 1001
Range Size : 1
Unique Internal Number + NO

DID number translator Identifier

External Number

Internal Number

Range of numbers

Step 8 Trunk group NPD selector:
The NPD number associated to this trunk has to be configured through the menu **Trunk Groups > Trunk group NPD selector > "select a Trunk Group ID"**.

Trunk group NPD selector
…
Trunk Group ID : *106*
Instance (reserved) : 1

Public NPD ID : *33*
Private NPD ID : 0
Management Mode + Normal

public SIP Trunk (205 in the example)

parameter has to match the NPD identifier (33 in the example)

parameter has to be set to Normal

Step 9 Network Routing Table:
The Network Routing Table associated to SIP trunk has to be configured through the menu **Translator > Network Routing Table > "select a Network Number"**.

Network Routing Table
…
Network Number : *0*

Rank of First Digit to be Sent : 1
Incoming identification prefix : --------
Protocol Type + ABC_F
Numbering Plan Descriptor ID : 11
ARS Route list : -1
Schedule number : -1
ATM Address ID : -1
Network call prefix : --------
City/Town Name : --------------------
Send City/Town Name + False
Associated Ext SIP gateway : -1
Enable UTF8 name sending + True

parameter matching the parameter Remote Network in the Trunk Groups (0 in the example)

parameter has to be set to ABC_F

Step 10 Trunk COS:
Accessible through the menu **External Services > Trunk COS > "select a Trunk Group COS = 31 for instance"**.

Trunk COS
…
Trunk COS : 31

Connection COS : 0
Trunk Type + ABC_F
Signaling Type + Not Relevant
Waiting Guide + True
Overflow Timer on No Answer : 300
Overflow Timer on Waiting : 300

T2 T0 ABC-F ISDN Trunks

Timer T303 : 100
Timer T304 : 300

parameter matching the parameter Trunk COS in the Trunk Group (31 in the example)

Parameter has to be set to a value greater than 110 (>11s)

Timer T310 : *110*
Timer T313 : 40
Timer T305 : 40
Timer T308 : 40
Timer T309 : 900

Step 11 SIP Ext Gateway:
An external gateway is the internal representation of a remote proxy/gateway of an operator. It is possible to declare one or several gateways on the IPBX. The SIP external gateway has to be configured through the menu **SIP > SIP Ext Gateway**.

Note: the Registration ID field was previously filled for the first SIP External Gateway. It is obsolete now. That was a workaround allowing probing both Nominal and Backup proxies.

- SIP Ext Gateway 1 (to Nominal Proxy):
- RE-INVITE without SDP is missing

SIP Ext Gateway

…
SIP External Gateway ID : 1

Gateway Name : *sip.osp.comsip.osp.com*
SIP Remote domain : *sip.osp.com*
PCS IP address : ---------------------------------------
SIP Port Number : 5060
SIP Transport Type + UDP
Belonging Domain : *sip.osp.com*
Registration ID : *+33296084675*
Registration ID in P_Asserted + False
Registration timer : *3600*
SIP Outbound Proxy : *imspcf211gm.sip.osp.com*
Supervision timer : 0
Trunk group number : *106*
Pool Number : -1
Outgoing realm : *sip.osp.com*
Outgoing username : *+33296084675@sip.osp.com*

Outgoing Password : *****
Confirm : *****

Incoming username : --

Incoming Password : ----------
Confirm : ----------

RFC 3325 supported by the distant + True
DNS type + DNS A
SIP DNS1 IP Address : ---
SIP DNS2 IP Address : ---
SDP in 18x + False
Minimal authentication method + SIP None
INFO method for remote extension + False
Send only trunk group algo + False

enter the name of the remote access/proxy (SBC111 in the example)

enter remote domain name (sip.osp.com in the example)

port of SIP messages intended for the remote gateway/proxy

type of transport used by the gateway

Enter registration id present in From/To headers of Register (+33296084675 in the example)

Enter registration timer (3600 in the example)

parameter has to match the SIP trunk (106 in the example)

Enter username for authorization (in format +33ZABPQMCDU@domain)

Enter password for authorization

parameter has to match the SIP trunk (106 in the example)

PAI supported for outgoing calls / From: anonymous@anomymous.invalid

*SDP **mustn't** be present in 180 ALERTING sent by the OXE*

***no** authentication requested by the proxy*

To EMS + False
SRTP + RTP only
Routing Application + False
Ignore inactive/black hole + *False*
Contact with IP address + False
Dynamic Payload type for DTMF : 101
100 REL for Outbound Calls : Not Supported
100 REL for Incoming Calls : Not Supported
Gateway type + *Default*
Re-Trans No. for REGISTER/OPTIONS : 4
P-Asserted-ID in Calling Number + False
Trusted P-Asserted-ID header + True
Diversion Info to provide through + Diversion
Trusted From header + False

emission of dual codec configured in the trunk group for outgoing calls

parameter has to be configured to False

dynamic payload proposed in RFC 2833 / recommended value is ***101***

100REL is supported for outgoing calls and required for incoming calls in order to generate PRACK requests

Gateway type has to be set to default

Step 12 Numbering Discriminator:
Accessible through the menu **Translator > External Numbering Plan > Numbering Discriminator**.

Numbering Discriminator
…
Discriminator No. : *10*
Name : BIV_SIP

Discriminator number (10 in the example)

Discriminator Selector name (BIV_SIP in the example)

Step 13 Discriminator Rule:
Accessible through the menu **Translator > External Numbering Plan > Numbering Discriminator > Discriminator Rule > "select a Discriminator No."**.

Note: Several **Discriminator Rules** should be configured according to the type of outgoing calls dialed (national/international).

Example1. *Discriminator Rule for* ***National*** *calls:*

Discriminator Rule
…
Discriminator No. : *10*
Call Number : *0*
Area Number : 1
ARS Route List Number : *10*
Schedule Number : -1
Number of Digits : *10*

Discriminator number (10 in the example)

rule for the dialed number (0 in the example)

parameter has to match the ARS Route for SIP Trunking (10 in the example)

parameter has to match the exact number of awaited digits (10 in the example)

Example2. *Discriminator Rule for* ***International*** *calls:*

Discriminator Rule
…
Discriminator No. : *10*
Call Number : *00*
Area Number : 1

Discriminator number (10 in the example)

rule for the dialed number (0 in the example)

parameter has to match the ARS Route for SIP Trunking (10 in the example)

ARS Route List Number : *10*
Schedule Number : -1
Number of Digits : *255*

For international calls the numer of digits is not limited.

Example3. *Discriminator Rule for* ***Short Numbering*** *calls:*

Discriminator Rule
…
Discriminator No. : *10*
Call Number : *3*

Area Number : 1
ARS Route List Number : *10*
Schedule Number : -1
Number of Digits : *4*

Discriminator number (10 in the example)

rule for the dialed number (3 in the example for numbers 3XXX)

parameter has to match the ARS Route for SIP Trunking (10 in the example)

parameter has to match the exact number of awaited digits (4 in the example)

Step 14 Discriminator Selector:
When a user dial a number, the user matches an **Entity Number** in **User** parameters. The Entity Discriminator has to be configured through the menu **Entities > Discriminator Selector > "select an Entity Number"**.

Discriminator Selector
…
Entity Number : *100*
Instance (reserved) : 1

Discriminator 00 : *10*
Discriminator 01 : 0
Discriminator 02 : 0
Discriminator 03 : 0
Discriminator 04 : 0
Discriminator 05 : 0

entity of site (100 in the example)

parameter has to match a Numbering Discriminator (10 in the example)

Step 15 Prefix Plan:
Accessible through the menu **Translator > Prefix Plan > "Prefix instance number"**.

Note: Dialing by overlap is not allowed by SIP.

Prefix Plan
…
Number : 0

Prefix Meaning + ARS Prof.Trg Grp Seizure
Discriminator No. : *0*

parameter has to be set to ***ARS Prof.Trg Grp Seizure***

parameter has to match the parameter Entity Discriminator No. (0 in the example)

Step 16 ARS Route list:
Accessible through the menu **Translator > Automatic Route Selection > ARS Route list**.

ARS Route list
…
ARS Route list : *10*

SIP ARS Route list number (10 in the example)

ARS Route list name (BIV_SIP in the example)

Name : *BIV_SIP*
PIN Code + False

Step 17 ARS Route :
Accessible through the menu **Translator > Automatic Route Selection > ARS Route list > ARS Route > "select a ARS Route list"**.

ARS Route
…
ARS Route list : *10*
Route : *1*

Name : *BIV_SIP*
Trunk Group Source + Route
Trunk Group : *106*
No.Digits To Be Removed : 0
Digits To Add : ----------------------------
Numbering Command Tabl. ID : 2
VPN Cost Limit : 0
Protocol Type + Dependant on Trunk Group Type
NPD identifier : 255
Route Type + Public
ATM Address ID : -1
Preempter + False
Quality
Quality + Speech

SIP ARS Route list number (1 in the example)
route of the SIP ARS (1 in the example)
ARS Route list name (BIV_SIP in the example)
parameter has to match the public SIP Trunk Group (106 in the example)
parameter has to match a SIP External Gateway
parameter has to be set to Speech and Fax

Step 18 Time-based Route List:
Accessible through the menu **Translator > Automatic Route Selection > ARS Route list > Time-based Route List > "select a ARS Route list"**.

Time-based Route List
…
ARS Route list : *1*
Time-based Route List ID : 1

Time-based Route

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

Time-based Route

Route Number : 1
Waiting Cost Limit : -1
Stopping Cost Limit : -1

SIP ARS Route list number (1 in the example)
route 1 to IMS

Step 19 Numbering Command Table: Must be moved before step 17
Accessible through the menu **Translator > Automatic Route Selection > ARS Route list > Numbering Command Table > "select a Numbering Command Table"**.

Numbering Command Table

Numbering Command Table (2 in the example)

```
…
Table ID : 2

Carrier Reference : 0
Command : ------------------------
Associated Ext SIP gateway : 2
```

*used by the accounting ticket / **'0"=not used***

parameter has to match the SIP gateway associated for SIP Trunking (2 in the example)

3.3 Voice Mail 4645

3.3.1 Payload 20ms (G711)

To comply with our requirement, payload 4645 for G711 has to be set to 20ms through the menu **IP > IP Parameters > G711 VOIP Framing for 4645 (only supported for Appliance Server and CS2).**

IP Parameters

…
System Option + G711 VOIP Framing for 4645

G711 VOIP Framing for 4645 + 20 ms

*G711 payload has to be set to **20ms***

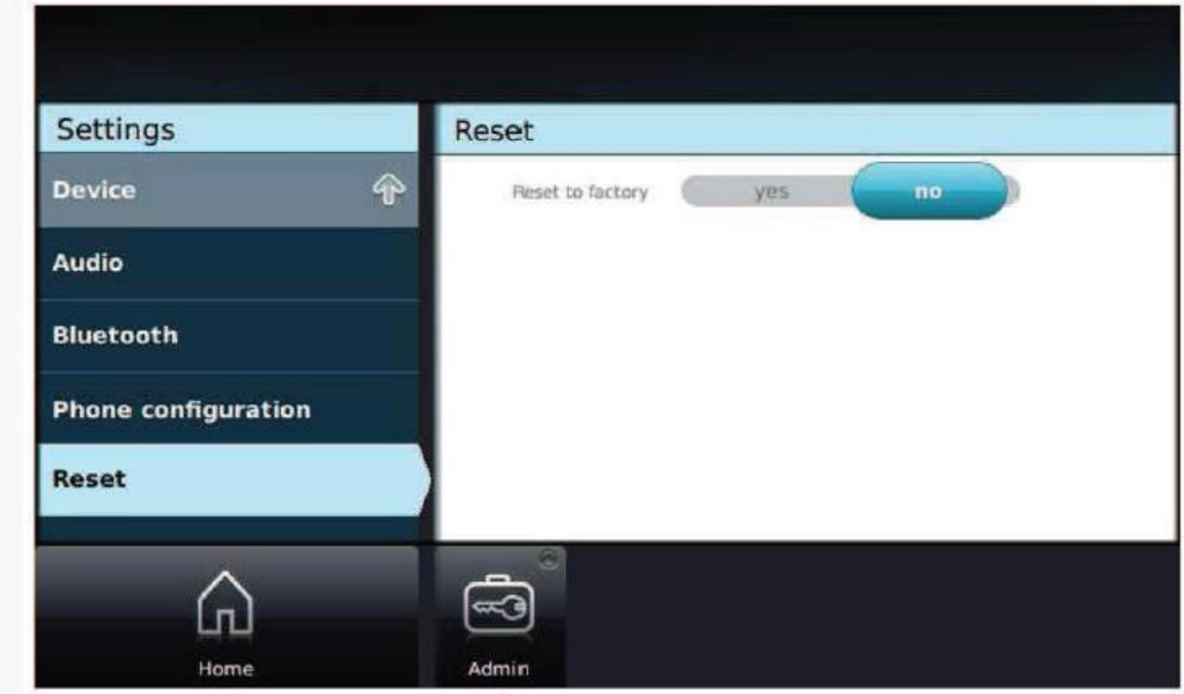
4 APPENDIX

4.1 8082 switching to NOE protocol

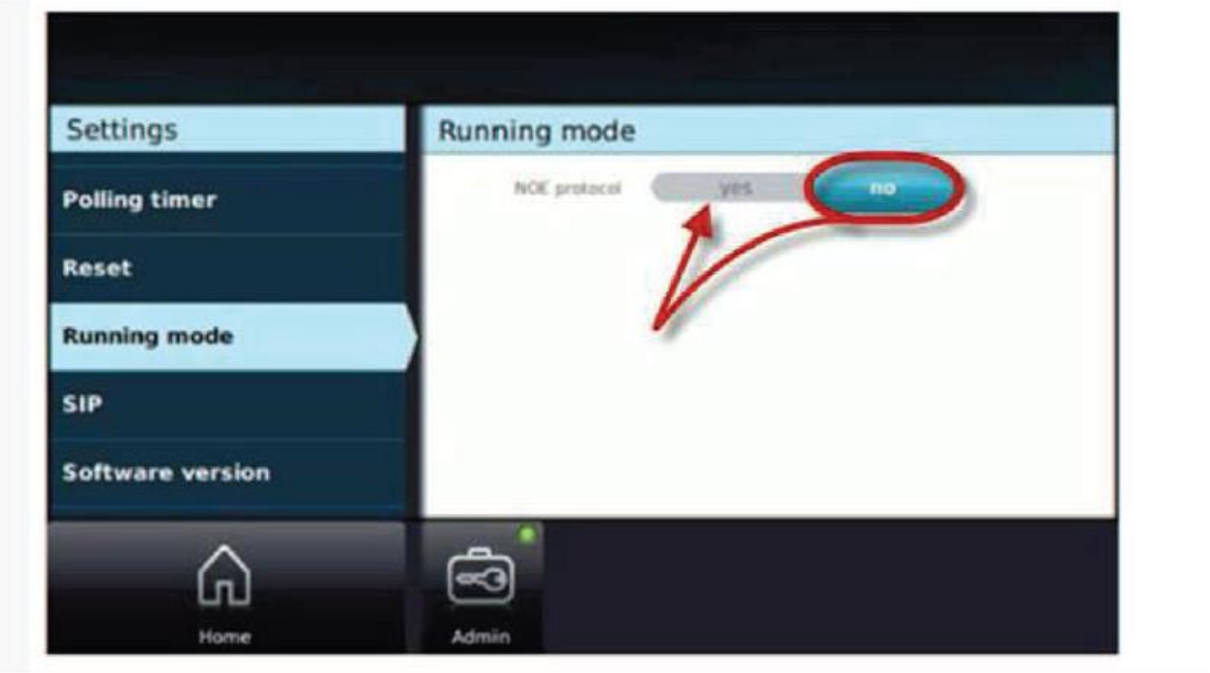
4.1.1 Protocol changing on a 8082 phone

This procedure is available since the firmware release R300-01.013.1. If the phone release is lower, see section 4.1.2 for upgrade.

1. On the VHE = Reset the factory in the VHE
2. Reboot your VHE
3. On the STEP 2 log you like Admin (0000)
4. In the Menu Device / drag the button to " reset to factory"



5. After reload wait step 2
6. On the STEP 2 log you like Admin (0000)
7. On the menu Device / running mode select the" Noe Protocle"



8. The VHE will be upgrade in the NOE mode (it take time)

4.1.2 8082 VHE upgrade to R300-01.013.1

NOTE! The release VHE-R300-01.013.1 or later must be delivered by Alcatel (it is no available through Business Portal). The upgrade procedure requires usage of ALEDS tool. Following instruction has been based on [3]

4.1.2.1 ALEDS tool

This tools allows to create a virtual machine including DHCP, http, DNS servers. This virtual machine can then be used to deploy binaries for a lot of Alcatel-Lucent Enterprise products (like OpenTouch, OmniPCX Enterprise, 8082 My IC Phone, ...).

It can be downloaded from the Enterprise Business Portal (Customer Support / Technical Support / Software download section / Alcatel-Lucent Enteprise Deployment Solution section).

4.1.2.2 Before disconnecting the 8082 My IC Phone from its previous system

Phone has to be put back to its "out of the box" defaults settings.

If the 8082 phone is connected to another system, connect to the phone using ssh and use embedded command "reset flash". In case the phone is in an old release like R200 for example, its CTL (Certificate Trusted List) has also to be deleted. To do that, use embedded command "CTL erase". Then do the "reset flash".

$ CTL erase
...
$ reset flash
...

If the phone is an "out of the box" one, these steps can be skipped.

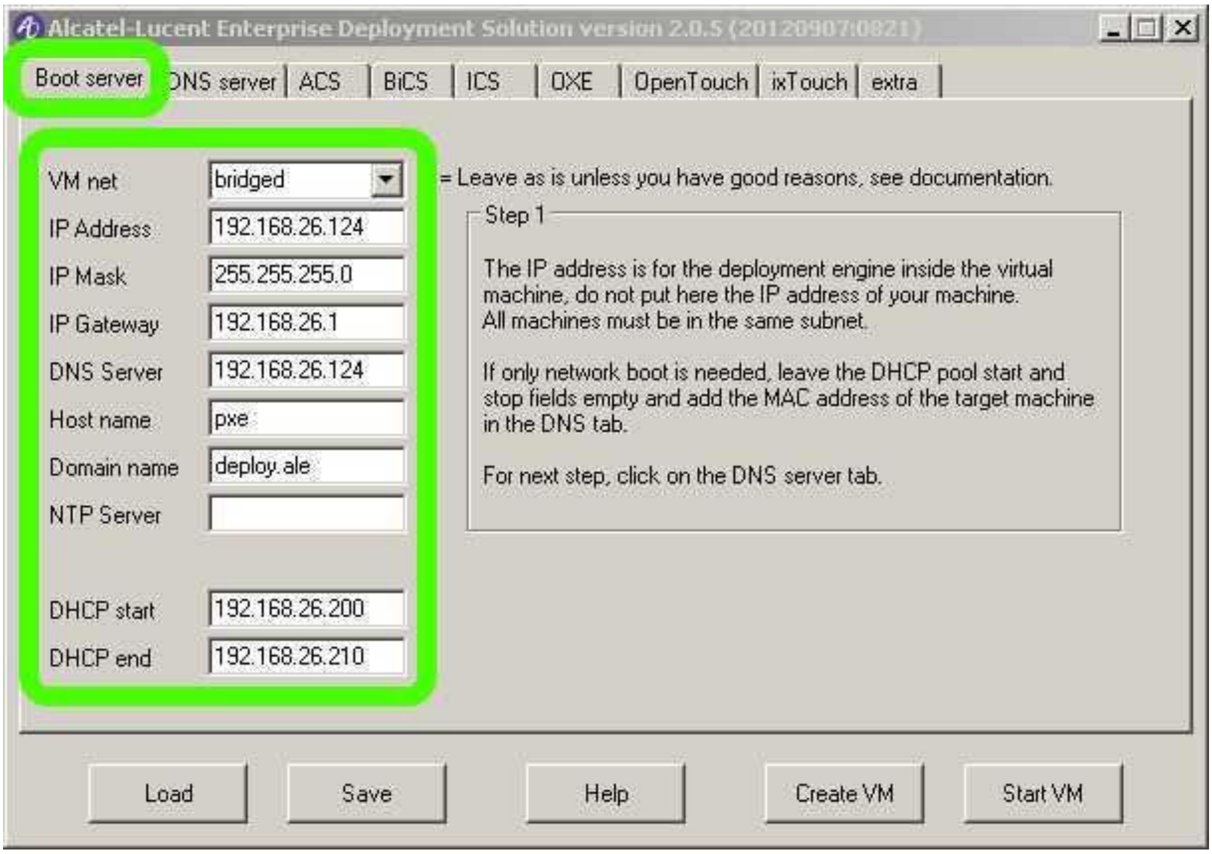
4.1.2.3 Install and configure the ALEDS tools

Once installed, check the two lines and modify them if needed the phone configuration file templates located in the tools root folder:

- ale-deploy-ict-cfg.in:

<setting id="DmSecucfgSsh" override="true" value="true" />
<setting id="DmAdminPasswd" value="0000" override="true"/>
Unzip the phone binary into a folder.
Then launch the tool and configure it as follows:

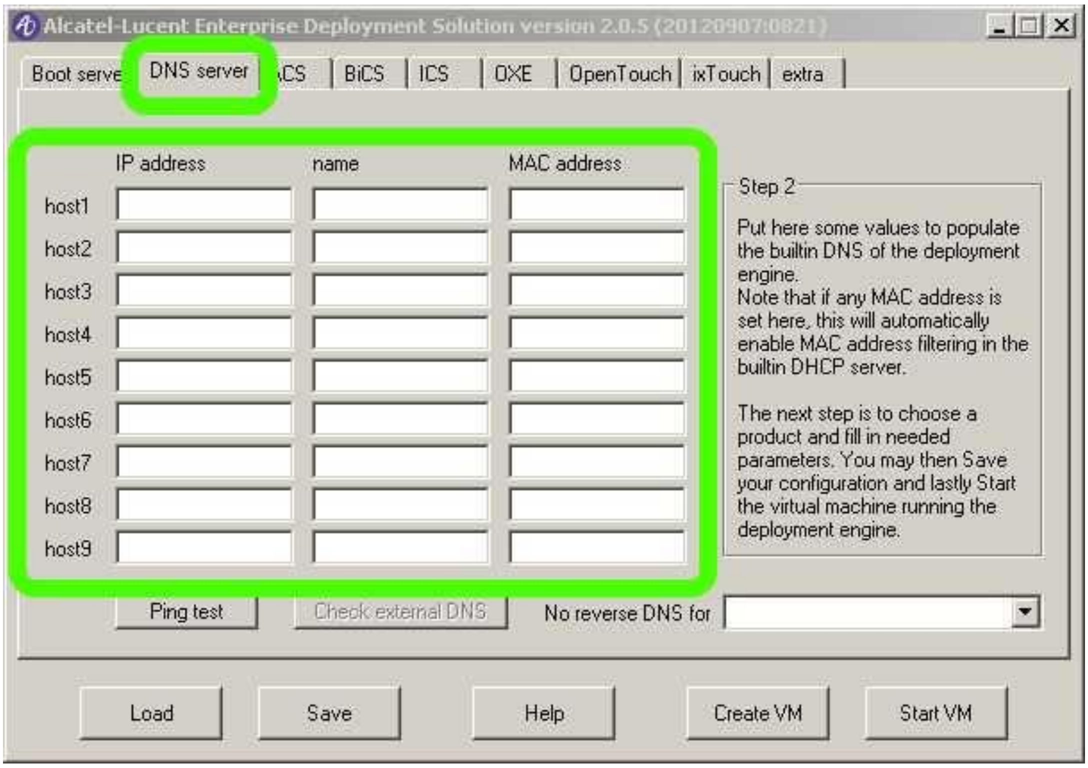
- Boot server parameters: configure the IP address, mask, gateway, DNS server, host name and domain name to fit your network.



Note that in the "bridget" mode, the Vitual Machine has to be configured to use an IP address fitting the real IP address used by the computer network card.

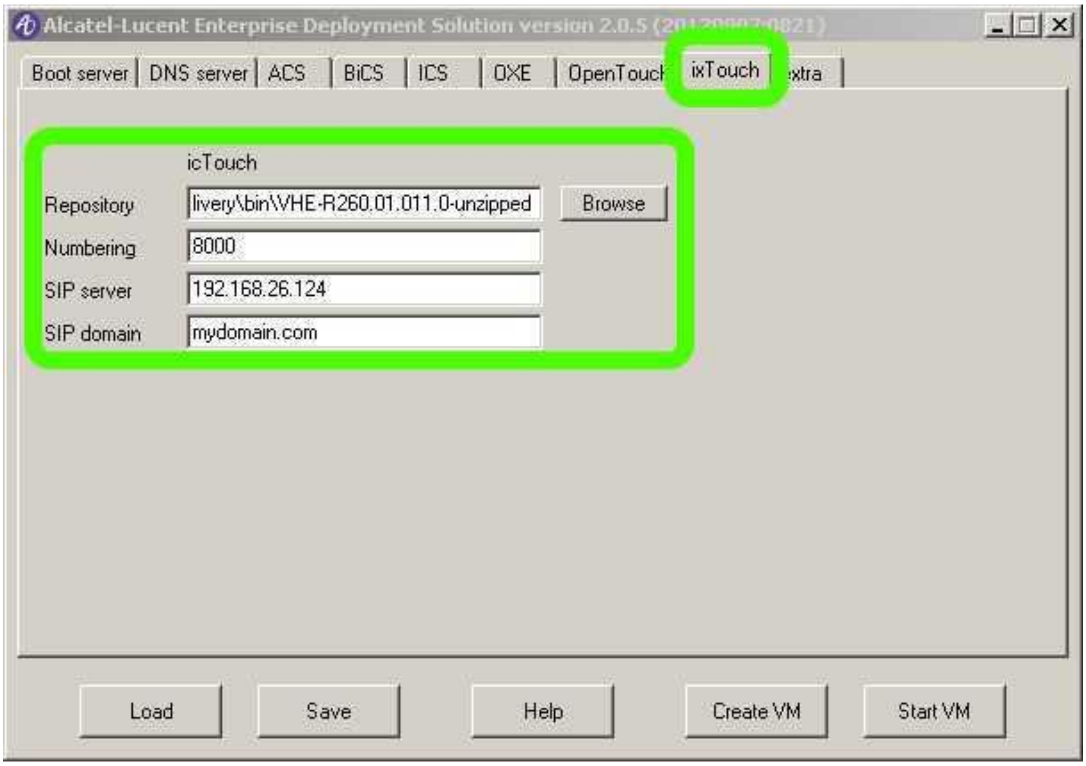
Host name and domain server parameters can be left to their default values.

- DNS server parameters: empty the MAC address table



- ixTouch: parameters used to handle the 8082 My IC Phone phones.

All these settings have to be filled in. If one is missing, the tool will not be able to handle the phones.

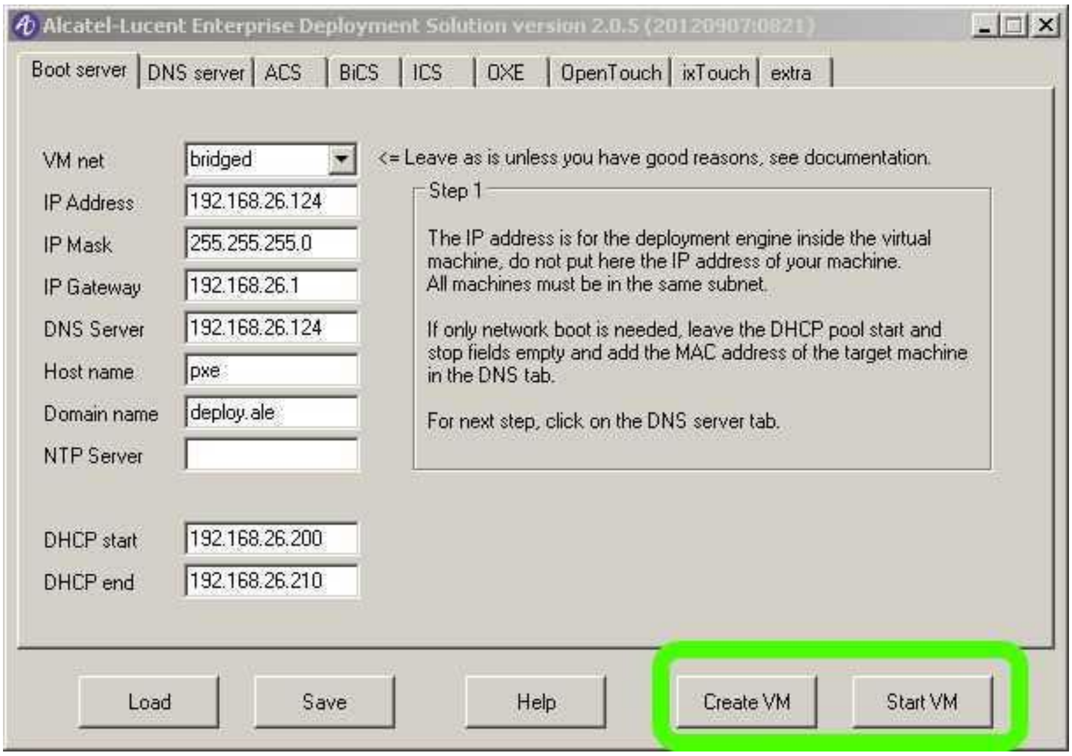


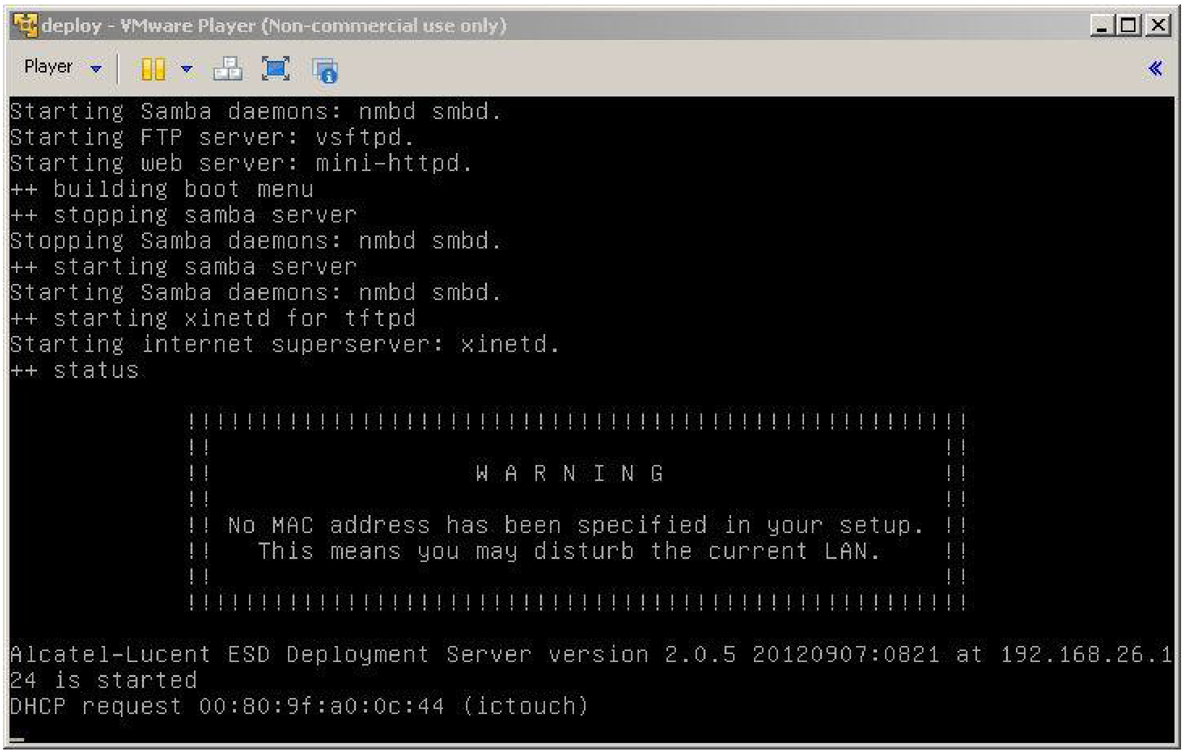
Repository: select the folder containing the UNZIPPED 8082 My IC Phone binary to install.
Numbering: enter whatever you want.
SIP server: enter whatever you want.

SIP domain: enter whatever you want.

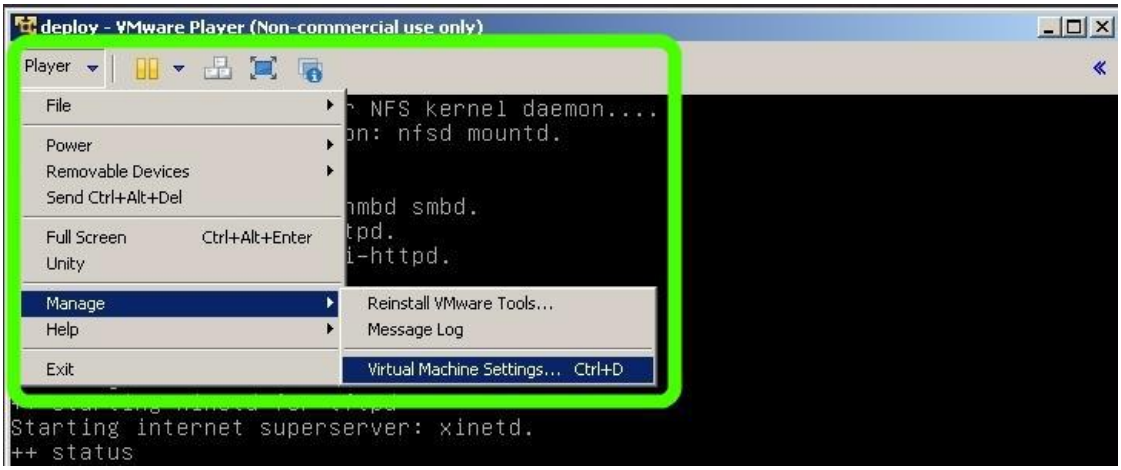
4.1.2.4 Upgrade / downgrade the phones

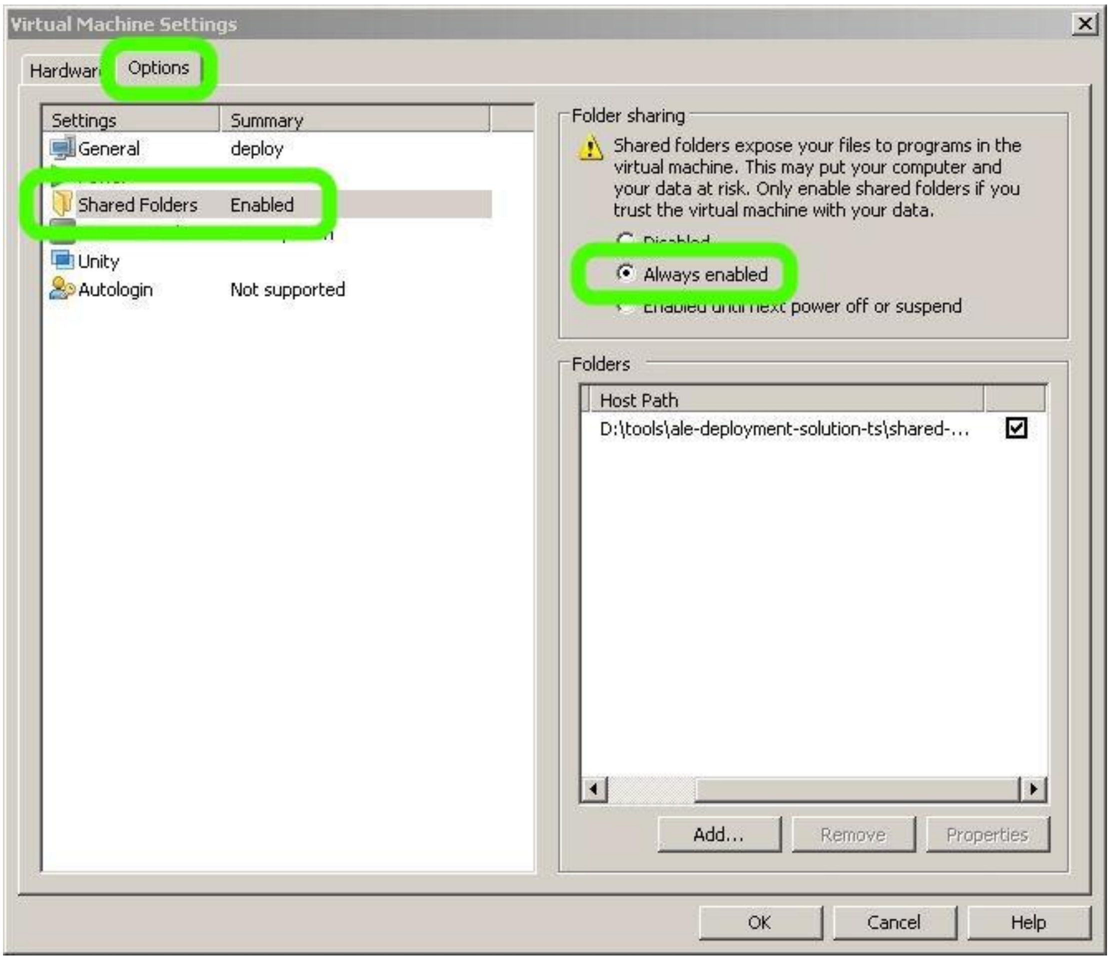
Start the virtual machine by using the "Start VM" button.





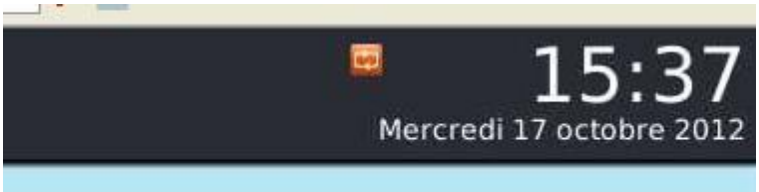
Check that the shared folders are enabled in the virtual machine settings:


deploy - VMware Player (Non-commercial use only)
Player
File
Power
Removable Devices
Send Ctrl+Alt+Del
Full Screen Ctrl+Alt+Enter
Unity
Manage
Help
Exit
Reinstall VMware Tools...
Message Log
Virtual Machine Settings... Ctrl+D
NFS kernel daemon....
on: nfsd mountd.
nmbd smbd.
tpd.
i-httpd.
Starting internet superserver: xinetd.
++ status

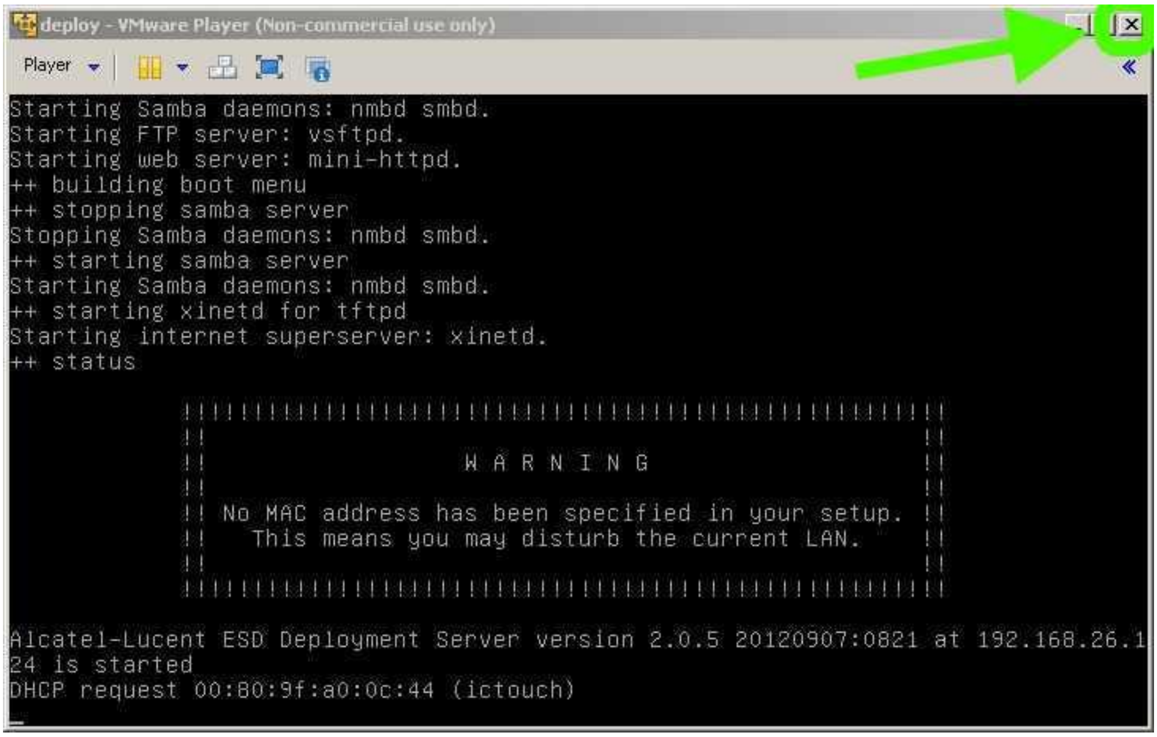


If not, enable them.
Connect your phone(s) to the network and wait for them to initialize themselves to the tool. They will get the IP parameters from the virtual machine DHCP server and then the configuration files from the virtual machine http server.
Then, the phone will start the binary upgrade.

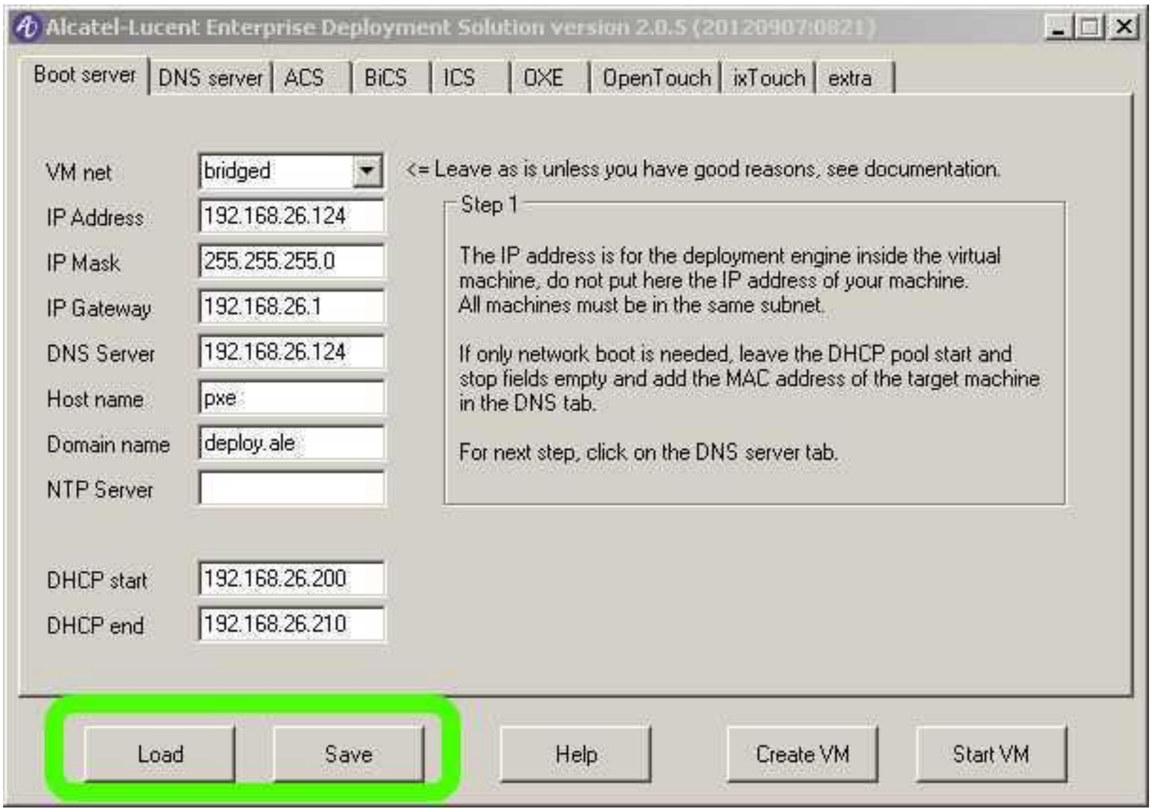
In case of it runs a R260 or more release, it will first download the rpms.
The phone will display the default home page and there will be an orange icon displayed in the top bar.



Once done, the phone will reboot and then install the downloaded rpms (and display the U "upgrade" screen).
Depending on the phone binary versions, these downloads and installations may last half an hour.
Once all the phones have been successfully upgraded, stop the virtual machine and stop the software deployment solution.



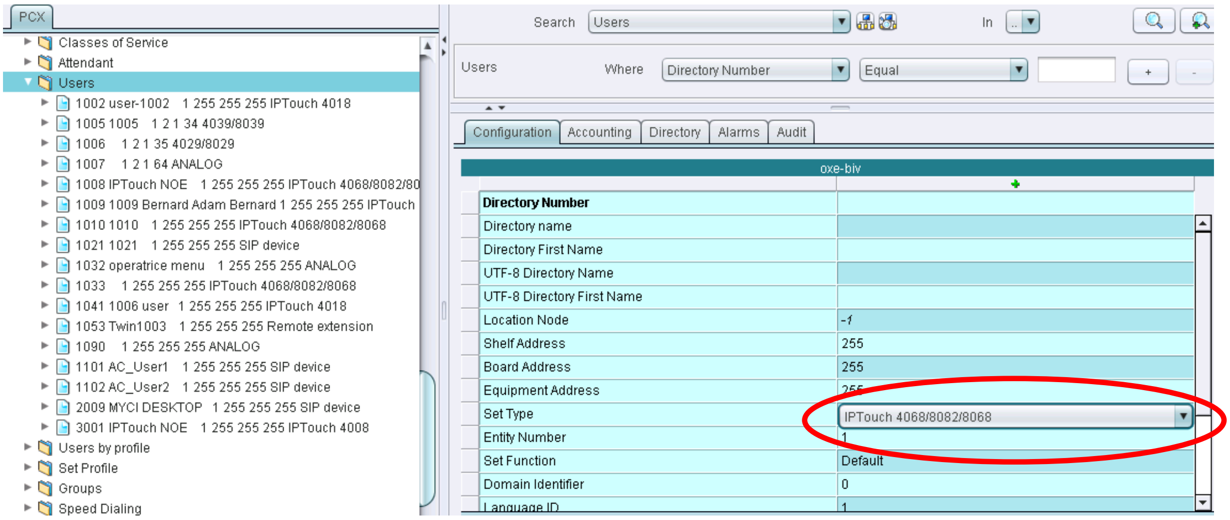
But do not forget to save your configuration before ("Save button") in order to be able to reuse it later ("Load" button).



Phone version can be checked by pressing the homepage "Settings" button and navigate to "Device / Software version".

4.1.3 Crating user on OXE for 8082 VHE with NOE

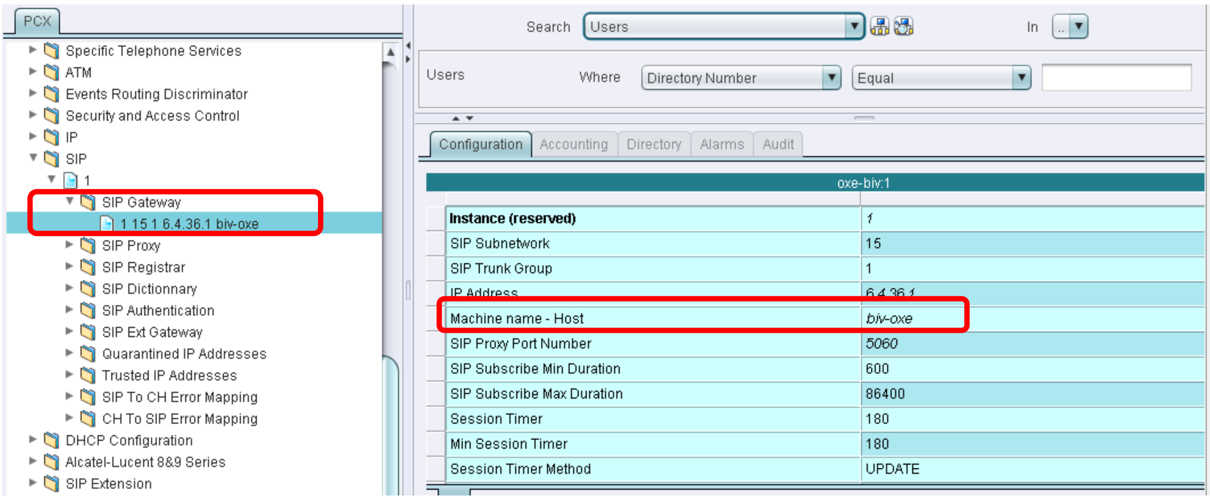
Add the new user with Set Type of “IP Touch 4068/8082/8068”



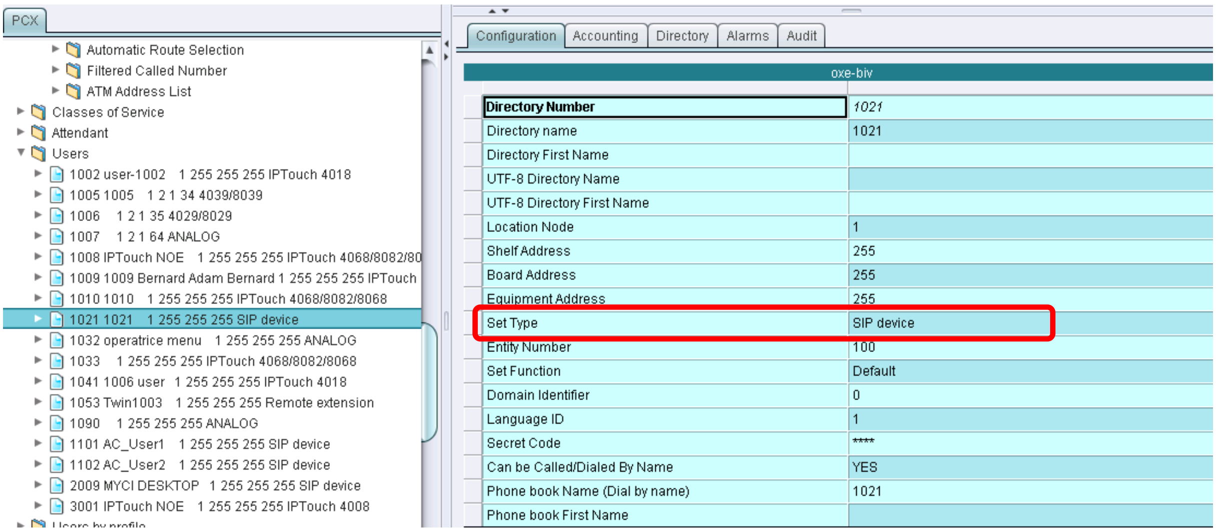
4.2 Conference module OT4135 configuration

4.2.1 OXE configuration

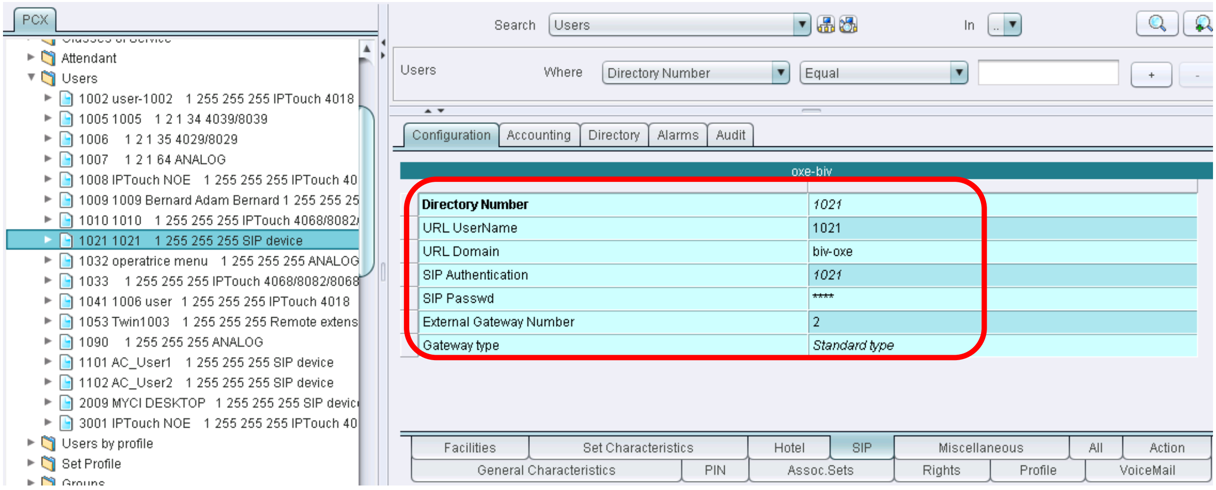
Check and remember “Machine name – Host” parameter value configured for SIP Gateway on OXE. In the example bellow it is “biv-oxe”.



Create a new user and set “Set Type” to “SIP device”

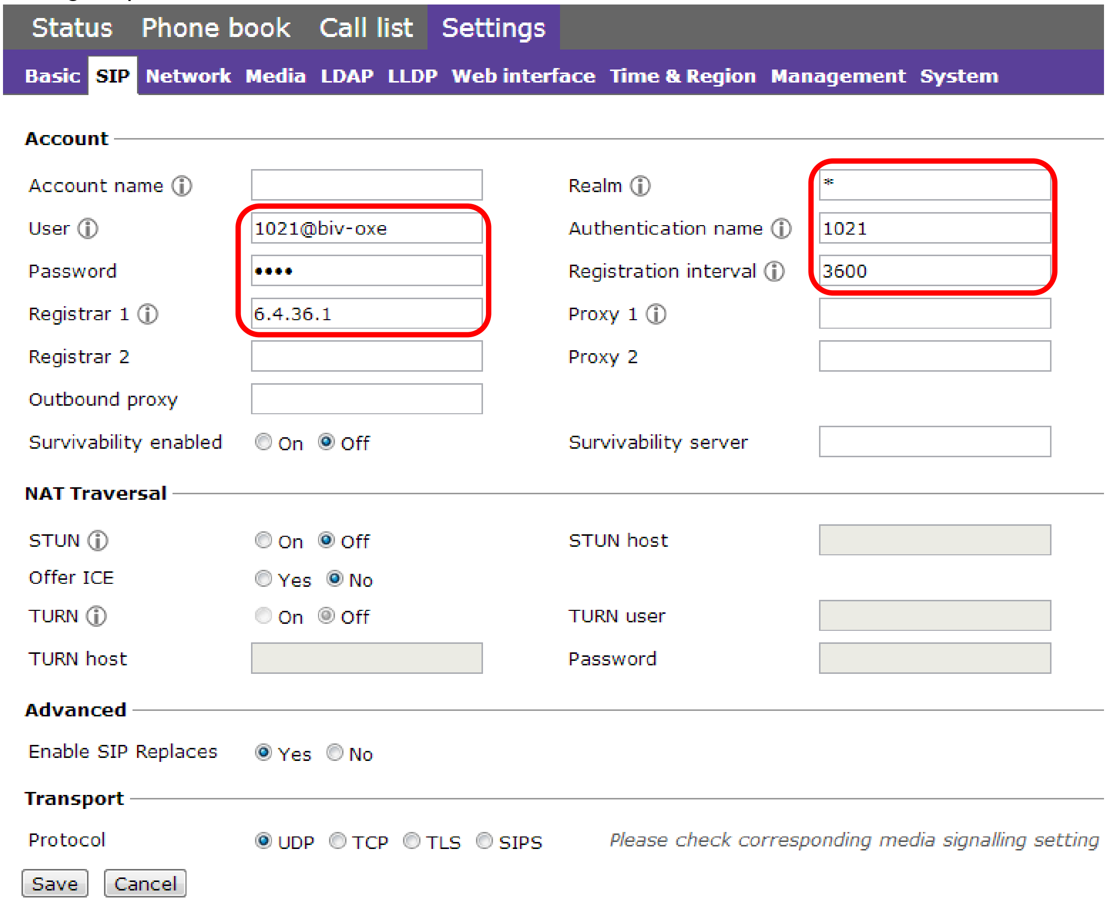


For the new user go to tab *SIP* and configure parameters as follow:



- URL Username: - user name
- URL Domain: *Machine name – Host* of the *SIP Gateway* on OXE
- Password
- External Gateway Number: – number of the *External Gateway* to BIV SIP network

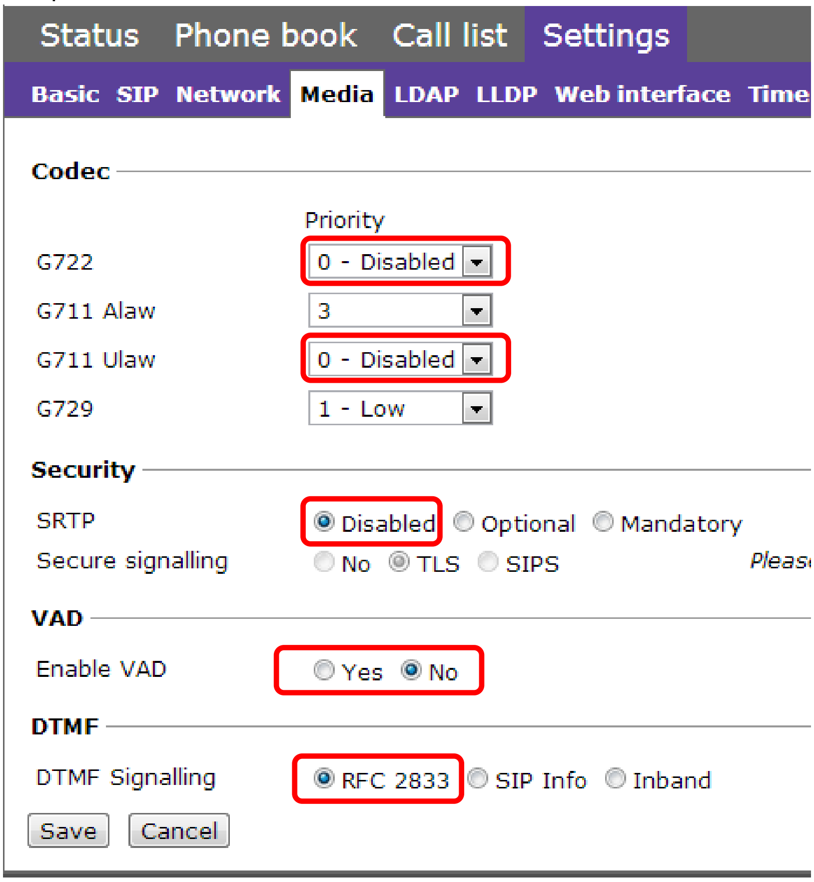
Log into OT4135 conference module through web browser. Go to *Settings>SIP*
Configure parameters as follow:



- User: *<ULR Username>@<URL Domain>*
- Password
- Authentication name : <SIP Authentication>

- Registration interval : 3600
- Registrar 1 : OXE IP address
- Outbound proxy : MUST leave **EMPTY**

Go to *Settings>Media*
Set parameters as follow:



- G722 : Disabled
- G711 Ulaw : Disabled
- Security : Disabled
- VAD : Disabled
- DTMF : RFC 2833

4.3 Attendant Console 4059 EE configuration

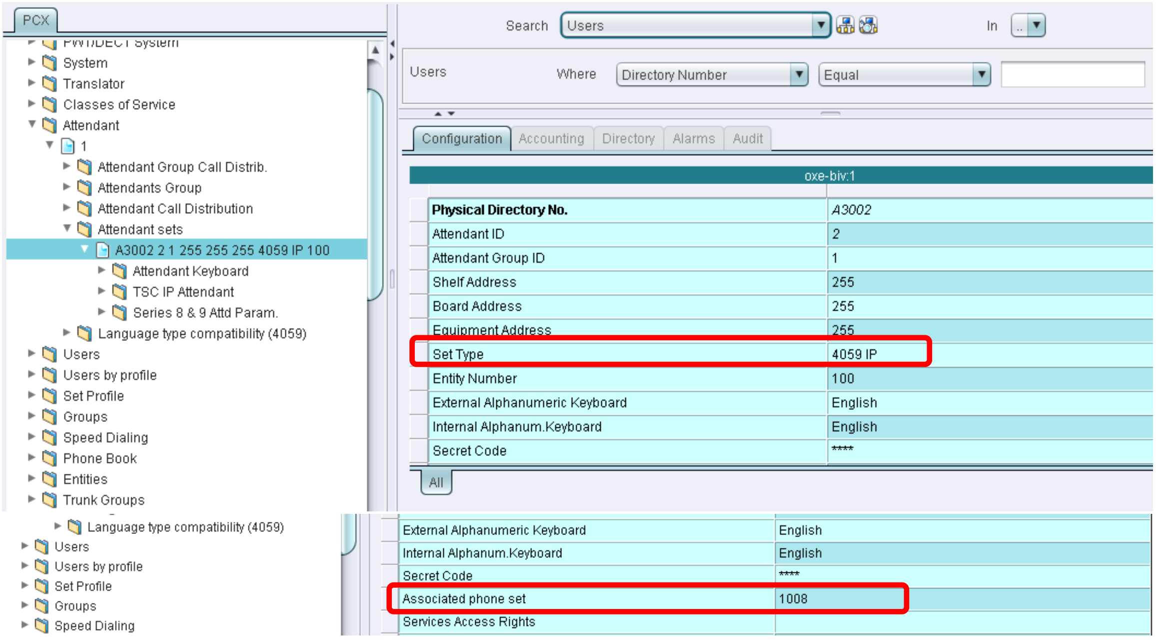
4.3.1 PC configuration requirements

For 4059 EE easy configuration without troubleshooting needs, make sure that the PC, on witch 4059EE is going to be installed, is equipped with **ONLY ONE NIC** card. Otherwise, if the PC has more than one nic and any problems occurs, refer to [4]

Deploy the PC within the same VLAN as OXE.

4.3.2 OXE configuration

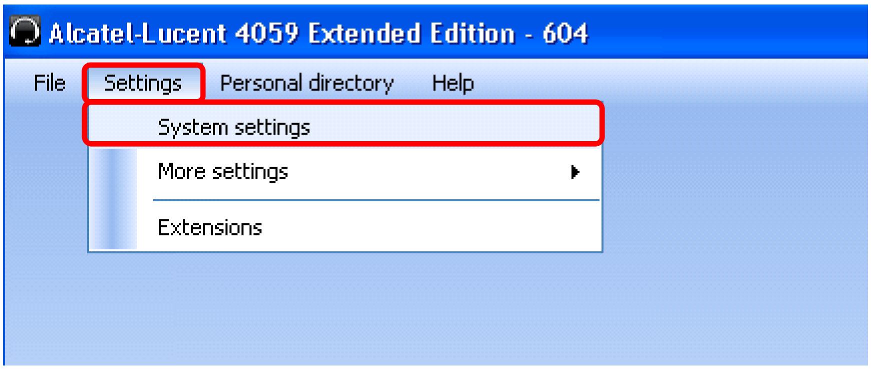
Create a new Attendant Console set.
Set parameters as follow



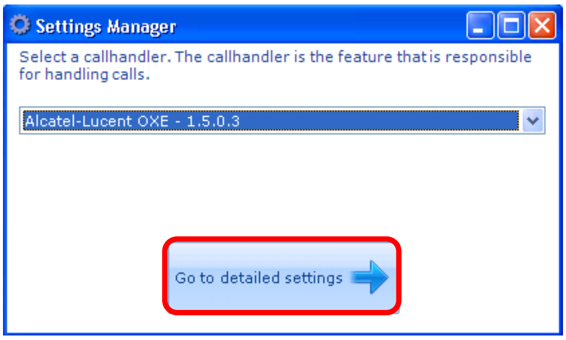
- Physical Directory No.: <Directory Number>
- Set type: 4059IP
- Associated phone set: DID of the Phone
-

4.3.3 4059 EE application configuration

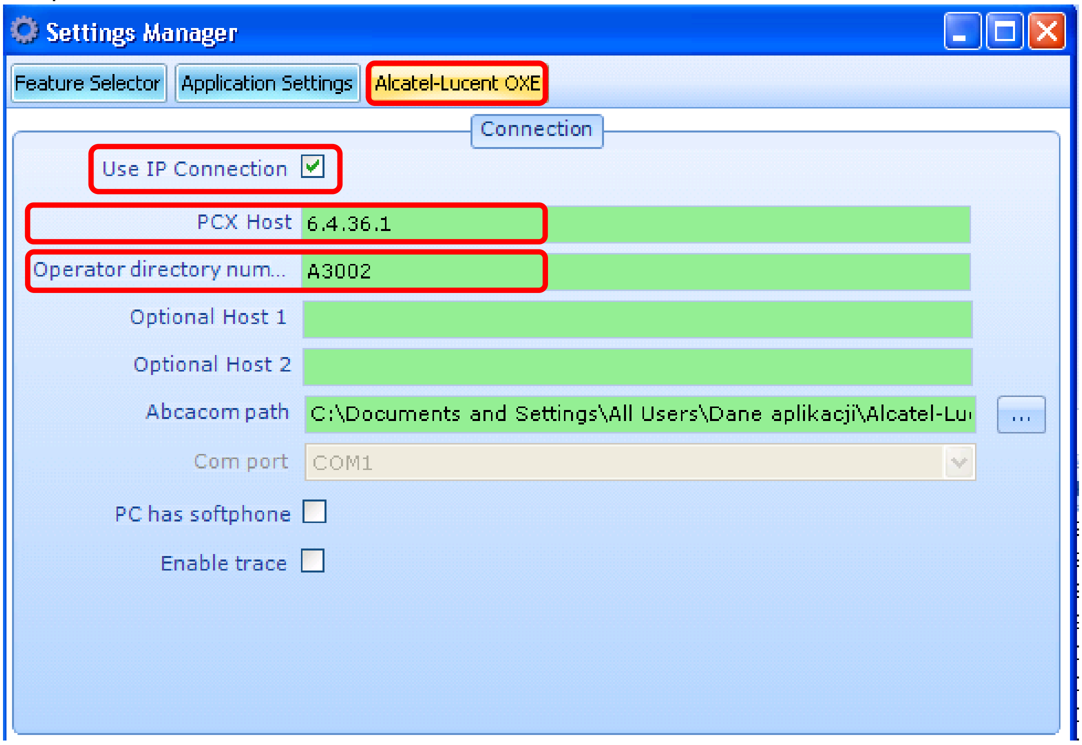
Run 4059EE application and go to *Settings>System Settings* menu.



Next *Go to detail settings*



Choose *Alcatel-Lucent OXE* tab.
Set parameters as follow:



- Use IP Connection : checked
- PCX Host: <OXE IP address>
- Operator direction num…: <Directory Number>