Technical Bulletin | Alcatel-Lucent OmniPCX Office

TC1975 ed.01 **Release 920**

SIP TRUNK SOLUTION: OPENIP (FR) CONFIGURATION GUIDELINE R920

This document details how to set up an OXO R920 IPBX for enabling a public SIP trunk of the Operator OPENIP in FRANCE.

Revision History

Edition 01: October 24, 2014 first R920 edition released for GA publication

Table of contents

1 General

This guide details "from-scratch" configuration of a R920 customer system featuring a SIP trunk of the Operator **OPENIP**. It is built under a particular perspective that ensures a convenient setup and correct operation of this SIP trunk.

1.1 References

Alcatel-Lucent documentation (available from the Business Partner Web Site):

[1] Alcatel-Lucent OmniPCX Office Communication Server- Expert Documentation

[2] Alcatel-Lucent OmniPCX Office Communication Server- Public SIP Trunking

1.2 Scope of the document

The doc is intended for normal-skilled engineers who are familiar with OMC and with the basic set up of the IPBX (e.g. complementary setup for local phones or for trunk resources is not considered). To be as universal as possible, it is based on the OMC tool used in English language.

When placed in a text area outside the OMC screen pictures, the config parameters /values can be easily identified in the document. They are typed in purple color with the heading sign ☞● . Some examples:

- ☞● **NP_International_Prefix = "00"** (quoted when value is freely edited in OMC)
- ☞● **VoIP_RTP_Direct = False** (non-quoted when value is selected from a user pick-list in OMC)
- ☞● **ARS_GWalive_Timer = (N/A)** (when the parameter is not shown or disabled in OMC)

Unless specific notification, the config values provided in this guide must be respected for all parameters whose name does not end with "**_example**". Indeed, the suffix "_example" is a mark of site-dependent parameters whose value needs to be customized. Examples:

- ☞● **LAN_Subnet_Mask_example = "255.255.255.0"**
- ☞● **Access_Channels_example = "4"**

As example values can be taken from a real customer site, some of them are masked or partially masked with asterisks for privacy reasons (i.e. local IP addresses, logins, public phone numbers). Examples:

- ☞● **NP_Instal_Number_example = "9****3430"**
- ☞● **GW_Reg_Username_example = "thJO******rUk"**

1.3 Scope of Alcatel-Lucent's support

The support delivered by Alcatel-Lucent is strictly delimited by the approved context and the system configuration detailed in this document. The protocol and functional aspects of the SIP trunk are in the scope, but not the audio quality of calls for the part incumbent on the Operator or linked to the client's IP infrastructure. Beyond these conditions, deployment of the solution on customer sites is supported under a specific service model: LA (Limited Availability) or GA (General Availability).

The up-to-date support model of an approved SIP trunk must be checked directly on Alcatel-Lucent's Web portal.

1.4 Software/ Hardware components on customer's infrastructure

INFRA COMPONENT	**MODEL**	**VERSION** (min compatible)
OXO IP-PBX system	Alcatel-Lucent OmniPCX Office	ALZFR920/056.001
OMC Management Application	Alcatel-Lucent OMC	OMC921/29.1a

1.5 Feature & Set Compatibility List

The tables 1.51 & 1.5.2 give the models of sets and the inter-op features that are agreed by ALU for the solution. Supported items are marked in columns "OK" (full support) or "WR" (With Restriction). Otherwise, non-supported items are shown in columns "NOK" (Not OK or Not Applicable) or "NT" (Not Tested).

For an item stated as "NT" (NOT TESTED):
In case you would like or need to have this item passed to the supported category and, if you are ready to collaborate with our approval process: simply send an email to "sip-for-smb@alcatel-lucent.com" and ask for the details related to the certification of the required item.

1.5.1 Supported Sets

RANGE OF SETS (OXO/OPENIP)	ALU SUPPORT		NO SUPPORT		REMARKS
	OK	WR	NOK	NT	
40x8 series	•				
40x9 series + Z + DECT				•	
IP-DECT DAP's				•	
4135				•	
My IC phone 8082	•				
8002/ 8012				•	
My IC mobile Android				•	
My IC mobile IPhone				•	
My IC & SIP Companion				•	

1.5.2 Supported Features

The following table lists the main inter-operation features supported.

FEATURE LIST (OXO/ OPENIP)	ALU SUPPORT		NO SUPPORT		REMARKS
	OK	WR	NOK	NT	
Approved Topology	•				*Remote Hosted Nat*
SIP Registration	•				*Provider WITH Registration*
SIP Authentication	•				*Provider WITH Authentication*
GW Dynamic Mgmt.	•				
Direct RTP			•		*Provider WITHOUT DIRECT RTP*
OutBound Basic Call	•				
InBound Basic Call	•				*Single codec (G711µ)*
Dynamic Codec Mgmt.			•		
Call Release	•				
Internal Call Transfer	•				
Internal Call Forward		•			
Call Hold	•				
InBound Call to DDI	•				
Reception of DTMF	•				
Emission of DTMF	•				
Emergency Calls	•				
Callers Repertory	•				
CLIP In. & OutBound	•				
CLIR In. & OutBound	•				
External Call Transfer	•				
External Call Forward		•			
Conference with 2 Ext.	•				
OutBound Fax T38				•	***Tested in G711***
InBound Fax T38				•	***Tested in G711***
OutBound Fax G711	•				
InBound Fax G711	•				
Dynamic Call Routing		•			
Busy State	•				
Preannouncement	•				

1.5.3 Restrictions

Internal Call Forward, External Call Forward and Dynamic Call Routing → Issue with COLP.

2 System General Info and Basic Setup

2.1 Pre-required information

The tables below show an example of all customized data that should be prepared in advance and available for starting OXO configuration. **The 2nd table gathers SIP data that must be supplied/ confirmed by the Operator** (e.g. designation of SIP proxies may evolve in time or differ upon geographic areas).

For both tables, if the "Example Value" is empty, the parameter is either not relevant or not mandatory for the configuration with this SIP Operator.

IP PARAMETERS

Data Type	Parameter	Example Value
IP Data subnetwork	OXO CPU - Data IP	192.168.137.246
	Default Gateway	192.168.137.254
	Netmask	255.255.255.0
	Data VLAN	
IP Voice Subnetwork	OXO CPU - Voice IP	192.168.137.246
	Default Gateway	192.168.137.254
	Netmask	255.255.255.0
	Voice VLAN	

Table of IP infrastructure parameters (to be customized for the customer site)

SIP TRUNK PARAMETERS

Data Type	Parameter	Example Value
Numbering Plan	Installation Number	9****3430
	Public DDI range	3431 3435
SIP-Trunk Data	SIP Operator GW IP	(N/A)
	Registered Username	thJO******rUk
	Authentication Login	thJO******rUk
	Password	*****
	SIP Operator Domain	sip9.voip-centrex.net
	SIP Customer Domain	
	SIP Realm	
	Outbound Proxy	sip9.voip-centrex.net
	Registrar IP address	
	Registrar name	
	Resolving DNS server 1	83.167.37.51
	Resolving DNS server 2	8.8.8.8

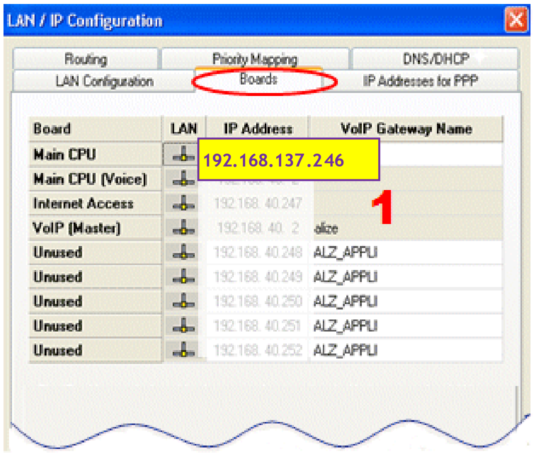
Table of SIP trunk parameters (communicated by the Operator)

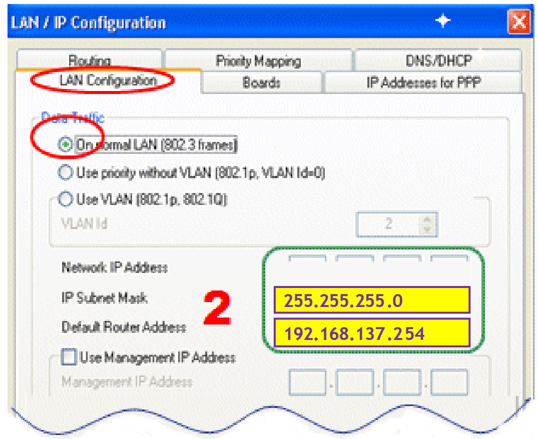
2.2 System Connection procedure

In order to configure the IPBX, it's necessary to use the OMC software at the Expert level session

2.3 Network configuration

The IP configuration is to be set before going further. This is done using the OMC menu Hardware and Limits -> LAN IP Configuration.





The "Boards" Tab shows the IP address for the main CPU and the default gateway. The main CPU IP address must be edited there ①

☞● **LAN_Main_CPU_IPadd_example = "192.168.137.246"**

The "LAN Configuration" Tab lets you edit the values of default gateway IP address and the subnet mask ②

☞● **LAN_Def_Router_IPadd_example = "192.168.137.254"**

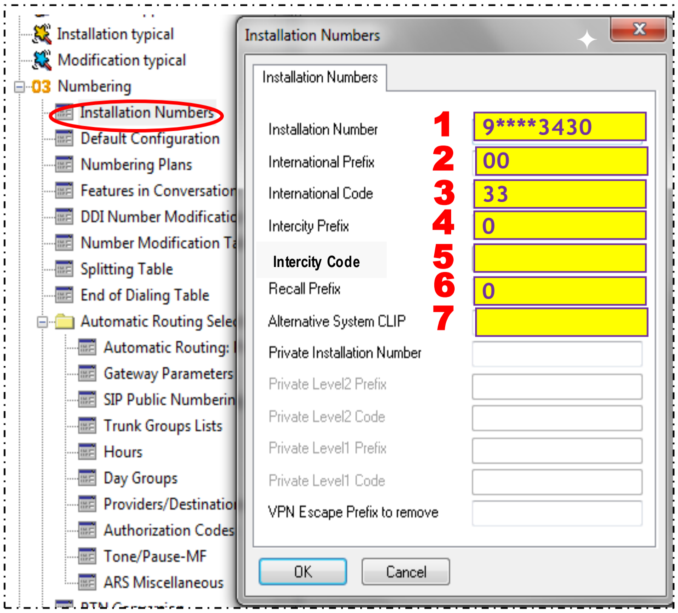
☞● **LAN_Subnet_Mask_example = "255.255.255.0"**

Once LAN-IP parameters are modified, a further system reboot ("Warm reset") is required to validate the changes!!

2.4 Numbering configuration

2.4.1 Installation numbers

As for any type of trunk, the public numbering used for SIP trunk is first ruled by the general numbering configuration of the PBX. Go to OMC Numbering -> Installation Numbers to check/ edit the "Installation Numbers" screen

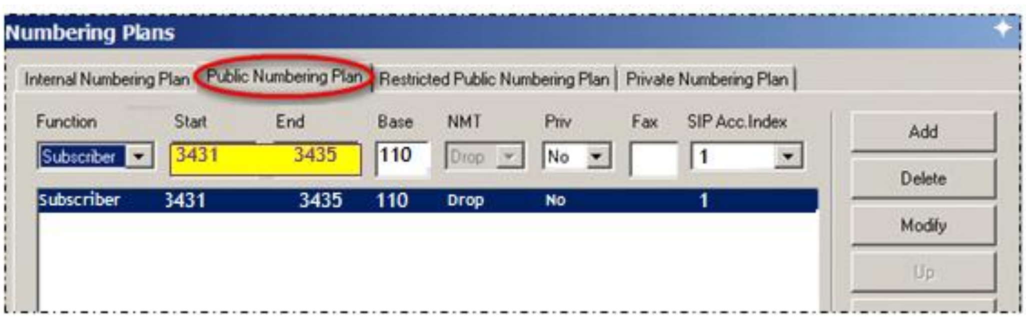


- Configuration for "Installation Numbers":

- ① Install. Number : ☞● **NP_Instal_Number_example = "9****3430"**
- ② International Prefix : ☞● **NP_International_Prefix = "00"**
- ③ International Code : ☞● **NP_International_Code = "33"**
- ④ Intercity Prefix : ☞● **NP_Intercity_Prefix = "0"**
- ⑤ Intercity Code : ☞● **NP_Intercity_Code_example = ""**
- ⑥ Recall Prefix : ☞● **NP_Recall_Prefix = "0"**
- ⑦ Alternative System CLIP : ☞● **NP_System_Alt_CLIP_example = ""**

2.4.2 DDI numbers

In OMC, the Public Numbering Plan is the place where to configure the DDI numbers allocated to the PBX subscribers. Go to OMC:.. Numbering -> Numbering Plans – Public Numbering Plan Tab...



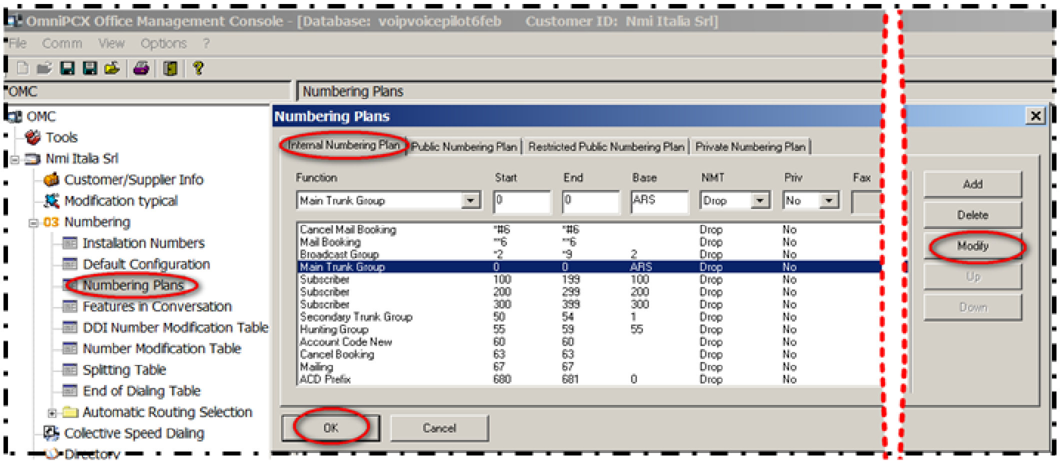
- Configuration for "Public Numbering Plan":

- ① DDI range : ☞● **NP_DDI_Range_example = "3431 3435"**

In conjunction with the configuration of section 2.4.1, this basic example allocates the DDI range "3431 3435" (i.e. 09****3431 - 09****3435) to the range of extensions "110 - 114"

2.4.3 Internal Numbering Plan

Accessible from OMC Numbering -> Numbering Plans menu, the internal numbering plan is the place where dialing of internal phones is first analyzed by the OXO system.



This example defines access to the internal ARS table for user dialing starting with digit 0. The "Drop" attribute indicates that the number dialed must be analyzed in the ARS Prefix table after dropping the initial digit 0.

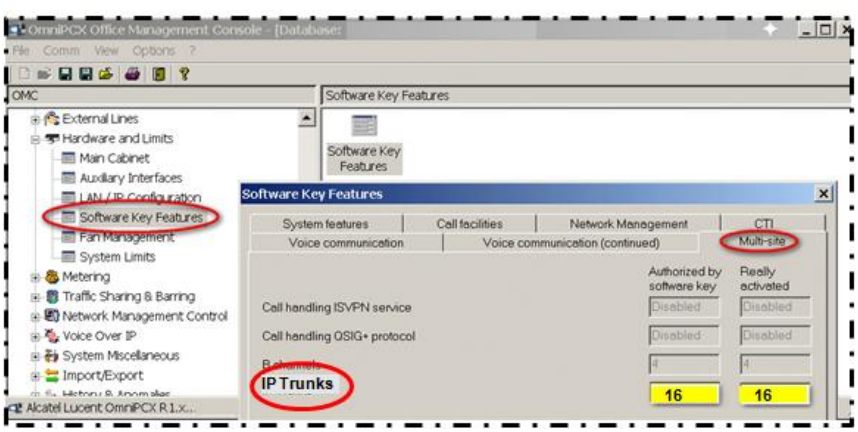
2.5 Traffic Sharing and Barring (reminder)

Though not described here, correct configuration of traffic sharing, barring and feature rights mechanisms is necessary to allow call features and outbound traffic over the SIP trunk.

3 Detailed SIP Trunk Configuration.

3.1 SIP ability check

A specific SW licence is mandatory to enable IP trunks on the system. The OMC menu Hardware and Limits -> Software Key Features gives an overview of the allowed features. In the Multi-site Tab, check the number of “IP Trunks” really activated (number of channels available for the VOIP trunk).

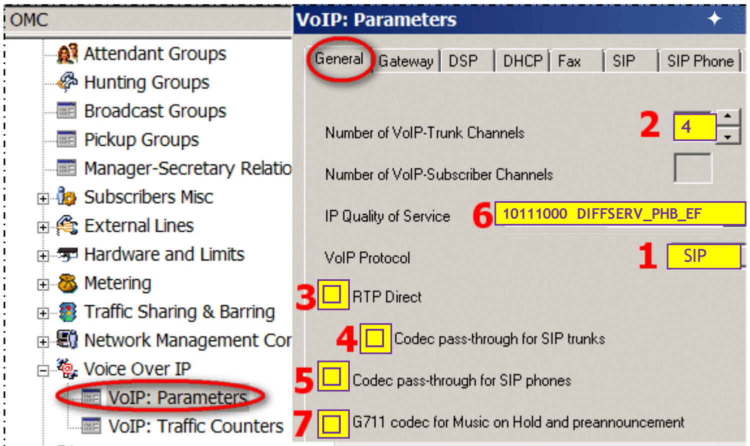


3.2 SIP Trunk creation

The VoIP trunk uses a specific signaling protocol (i.e. SIP) and physical resources embedded in the IPBX (i.e. DSP's). To create the SIP trunk in the system, it's first necessary to allocate some of the DSP resources to it.

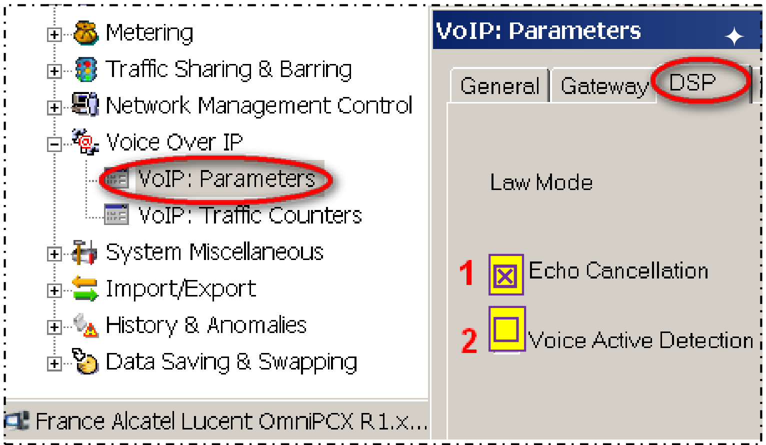
3.2.1 Management of VoIP channels

Go to OMC Voice Over IP -> VOIP:Parameters - General Tab to check/ edit the requested parameters.



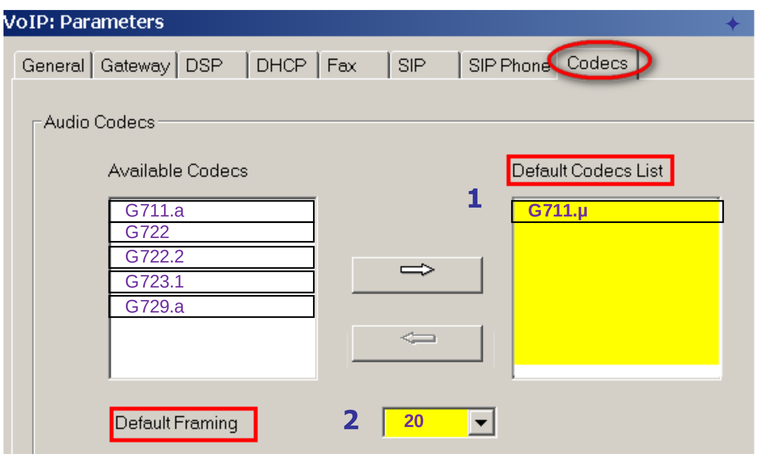
- ① the VOIP protocol in use :
  - ☞● **VoIP_Protocol = SIP**
- ② from the amount available in the system, the number of channels dedicated to VoIP trunks:
  - ☞● **VoIP_Trunk_Channels_example = "4"**
- the RTP routing mode and codec management for IP extensions:
  - ③ "RTP Direct" :
    - ☞● **VoIP_RTP_Direct = False**
  - ④ "Codec pass-through for SIP trunk":
    - ☞● **VoIP_Trunk_Codec_Passthru = False**
  - ⑤ "Codec pass-through for SIP phones":
    - ☞● **VoIP_Phone_Codec_Passthru = False**
- the IP Quality of Service :
  - ⑥ "IP Quality of Service" :
    - ☞● **VoIP_IP_QoS_example = 10111000 DIFFSERV_PHB_EF**
- the control of G711 codec for Music On Hold:
  - ⑦ "G711 codec for MOH and preannouncement " :
    - ☞● **VoIP_G711_MOH = False**

With OMC menu Voice Over IP -> VOIP:Parameters - DSP Tab, also define the global control of Echo cancellation and the VAD parameters of VoIP calls.



- Set DSP parameters as follows:
  - ① "Echo Cancellation" : ☞● **VoIP_DSP_Echo_Cancel = True**
  - ② "Voice Active Detection" : ☞● **VoIP_DSP_VAD = False**

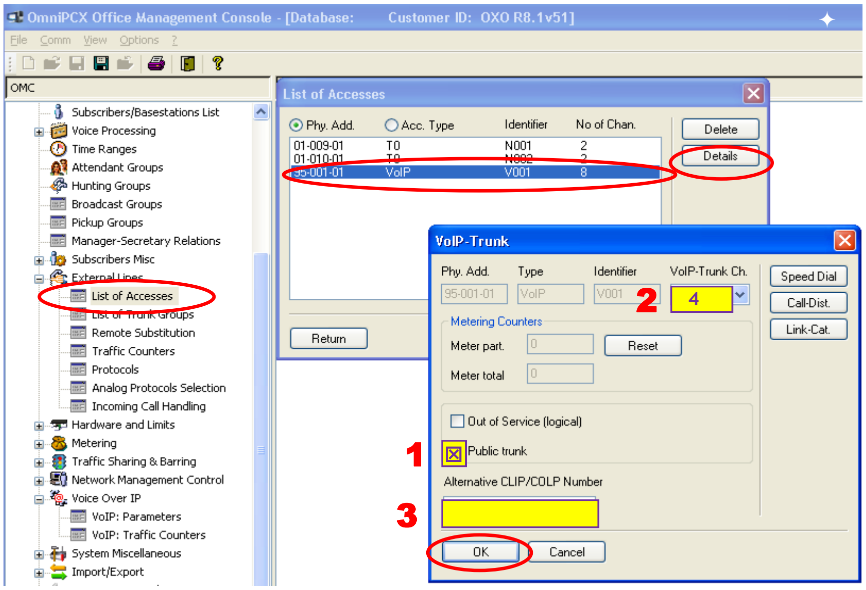
With OMC menu Voice Over IP -> VOIP:Parameters - Codecs , use the Right/ Left arrows to set up the customized "Default Codecs List" (yellow zone). Also define the "Default Framing" (in ms) valid for this codecs list.



- Set Codecs tab as follows:
  - ① "Default Codecs List" :
    ☞● **VoIP_Def_CodecList = G711.μ**
  - ② "Default Framing" :
    ☞● **VoIP_Def_CodecFraming = "20"**

3.2.2 "Physical" Access associated to the SIP Trunk

From OMC External Lines -> List of access, select the VoIP access associated to the VoIP trunk.



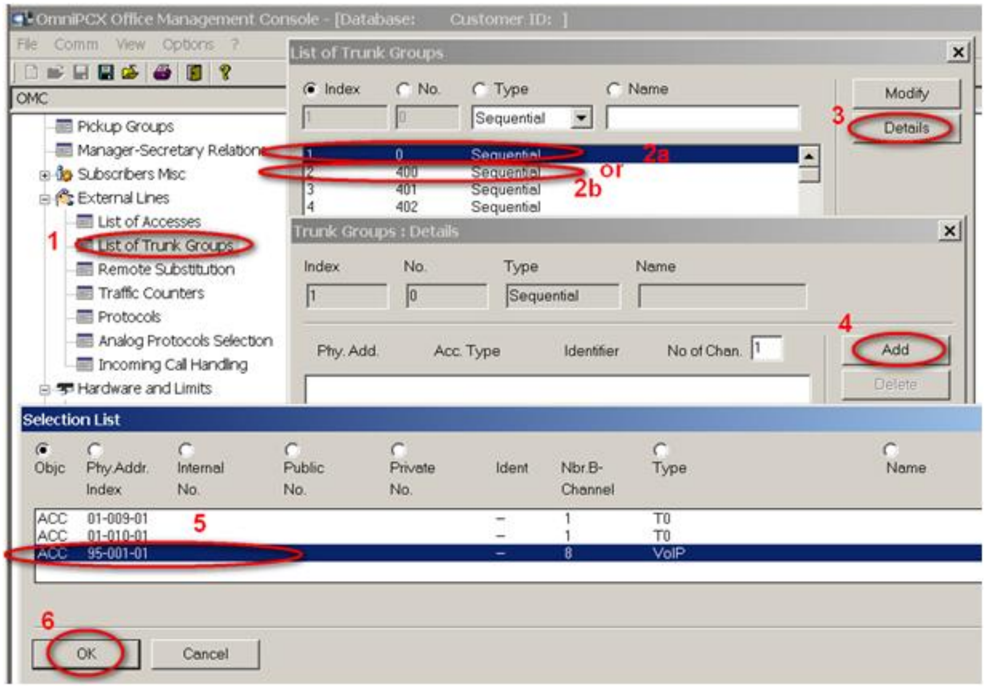
Then, configure the access parameters:

- ① "Public trunk" option : ☞● **Access_is_Public = True**
- ② number of channels allocated to the SIP trunk access "VoIP-Trunk Ch.":
  ☞● **Access_Channels_example = "4"**
- ③ "Alternative CLIP/COLP Number" :
  ☞● **Access_Alt_CLIP_example = ""**

3.2.3 Hosting System Trunk Group

To enable phone calls over the SIP trunk, it's necessary to have this latter included within one Trunk Group of the system. Two alternative cases are considered here below.

Select the OMC menu External Lines ->List of Trunk Groups and carry out the selections and push-button steps shown in the following picture (i.e. step1 to step 6 depicted by red digits 1 to 6).



As a configuration variant, at step 2 you can include the SIP trunk access into the OXO's main Trunk Group (i.e. step 2a for index #1) or into one of the secondary Trunk Groups (e.g. step 2b for index #2).

The SIP trunk can be placed freely into one or several Trunk Groups of the system: several Trunk Groups permits to manage traffic sharing based on called destinations and/or access rights assigned to internal subscribers.
The index number selected at step 2 is used in further configuration at section 3.5

3.3 ARS context associated to the SIP Trunk

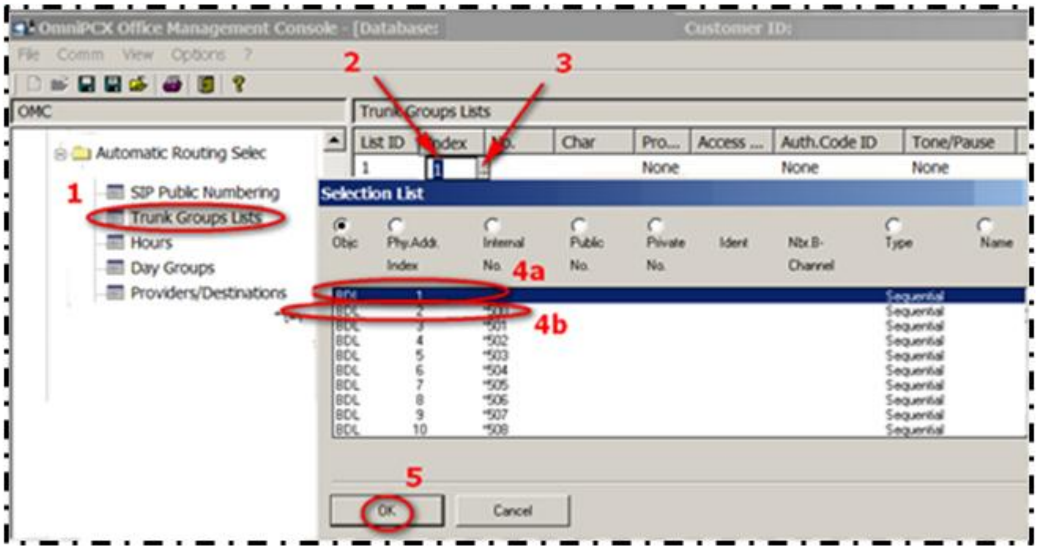
In-depth configuration of the SIP trunk is carried out via the ARS context associated to the trunk and is documented in the following sub-chapters. One key parameter of this configuration is the IP address of the peering SIP Gateway located on the Operator's network. OXO permits to select between static mode (i.e. fixed IP) or dynamic mode (i.e. automatic IP address resolution carried out by OXO). The configuration retained depends on the Operator's capabilities.

This SIP Operator does provide service for dynamic IP address resolution of its SIP Gateways and therefore, configuration associated to DNS-SRV is applied in the following ARS sections.

3.3.1 ARS Trunk Groups Lists

To enable voice calls via the ARS system, it's necessary to have ARS Trunk Groups created via the OMC menu Numbering -> Automatic Routing Selection -> Trunk Groups Lists.

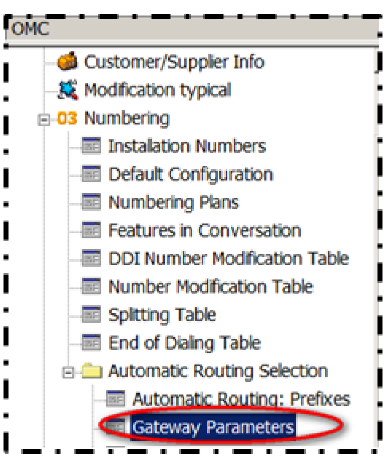
In this menu, new lines are created after clicking the mouse right button and selecting function "Add"



Carry out the selections and push-button steps 1 to 5 above. At step 4, you need to select the line index corresponding to a System Trunk Group previously defined at section 3.2.3 (i.e. selecting 4a for the Main Trunk Group or 4b for the secondary Trunk Group of index #2).

3.3.2 ARS Gateway Parameters

The OMC menu "Numbering" → "Automatic Routing Selection" → "Gateway Parameters" gathers different parameters proper to the peer SIP gateway. These inputs include IP access (e.g. proxy, domain name, address resolution, sip port, ...), account registration, authentication (till OXO R910), fax mode (T38 or G711 pass-thru).



The following SIP Gateway parameters ["GW_Target_Domain", "GW_Outb_Proxy", "GW_Reg_Name", "GW_Prim_DNS", "GW_Sec_DNS"] are controlled by the Operator and might vary in time or upon other own criteria. In case of any doubt, those values must be revised directly with the Operator.

In the present configuration, the SIP trunk is conveyed by the gateway referred as index 1. For this Gateway line, set up the whole parameters as shown in the three partial pictures below (i.e. left part + middle part + right part):

Gateway Parameters **(Left part)**

Index	Login	Passw...	Domain Name	Realm	RFC 3325	Remote SIP Port	SIP Numbers Format I...
1			sip9.voip-centrex.net		Yes	5060	1
	1	2	3	4	5	6	7

- ① "Login" : **Since R920, must be configured in "ARS SIP Accounts"**
- ② "Password" : **Since R920, must be configured in "ARS SIP Accounts"**
- ③ "Domain Name" : ☞● **GW_Target_Domain = "sip9.voip-centrex.net"**
- ④ "Realm" : ☞● **GW_Realm = ""**
- ⑤ "RFC 3325" : ☞● **GW_RFC3325 = Yes**
- ⑥ "Remote SIP Port" : ☞● **GW_Remote_SIP_Port = "5060"** (see Note below)
- ⑦ "SIP Numbers ..." : index number 1 shown as example (detailed at Ch. 3.5.5)

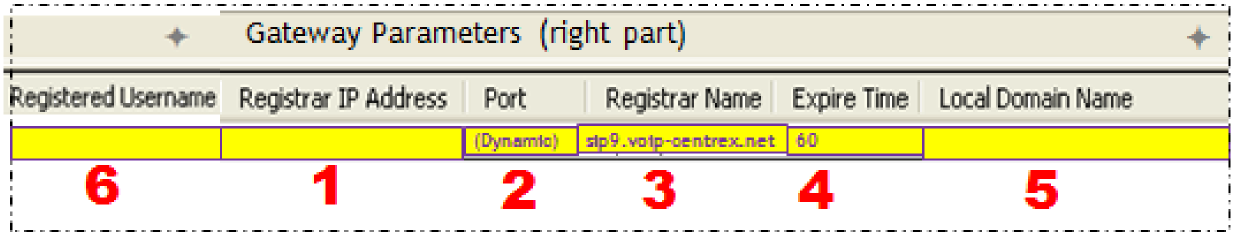
Default value of Remote SIP port is 5060. After completing the further parameters, this value will automatically change to "Dynamic" whenever the parameter DNS is set to "DNS SRV".

Gateway Parameters **(middle part)**

DNS	Primary DNS Server	Secondary DNS Server	Outbound Proxy	Fax	Index Label	Registration Requested	Public DDI Registrat
DNSSRV	83.167.37.51	8.8.8.8	sip9.voip-centrex.net	G711		Yes	No
1	2	3	4	5		6	7

- ① "DNS" : ☞● **GW_DNS_Mode = DNSSRV**
- ② "Primary DNS..." : ☞● **GW_Prim_DNS = "83.167.37.51"**
- ③ "Secondary DNS" : ☞● **GW_Sec_DNS = "8.8.8.8"**
- ④ "Outbound Proxy" : ☞● **GW_Outb_Proxy = "sip9.voip-centrex.net"**
- ⑤ "Fax" : ☞● **GW_Fax_Mode = G711**
- ⑥ "Registration Req." : ☞● **GW_Reg_Requested = Yes** (see note after)
- ⑦ "Public DDI Registr." : ☞● **GW_Reg_PubDDI = No**

- While "GW_Reg_Request " is equal to "No", the remaining parameters related to registration get disabled in OMC.
- The "Registrar IP Address" is automatically set to blank if DNS parameter is set to "DNS SRV".

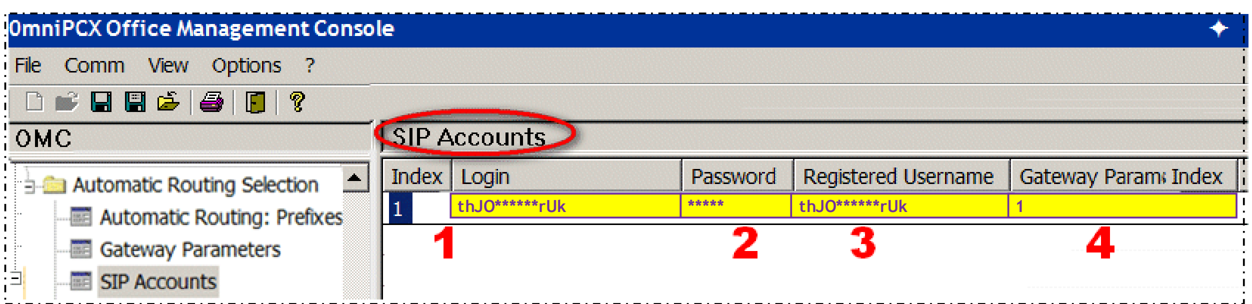


- ① "Registrar IP .." : ☞● **GW_Reg_IP_Address = (N/A)** (see Note after)
- ② "Port : ☞● **GW_Reg_Port = "(Dynamic)"** (see Note after)
- ③ "Registrar Name" : ☞● **GW_Reg_Name = "sip9.voip-centrex.net"**
- ④ "Expire Time" : ☞● **GW_Reg_Expire_Time = "60"**
- ⑤ "Local Domain .." : ☞● **GW_Local_Domain_Name = ""**
- ⑥ "Registered Usern." : **Since R920, must be configured in "ARS SIP Accounts"**

When "DNS-SRV " selection is made for previous gateway parameter "DNS", then the above items ① and ② become read-only: the actual values are fixed by the Operator thru automatic/ dynamic resolution (those displayed just reflect the temporary in use when the system configuration was last read out/ saved by OMC).

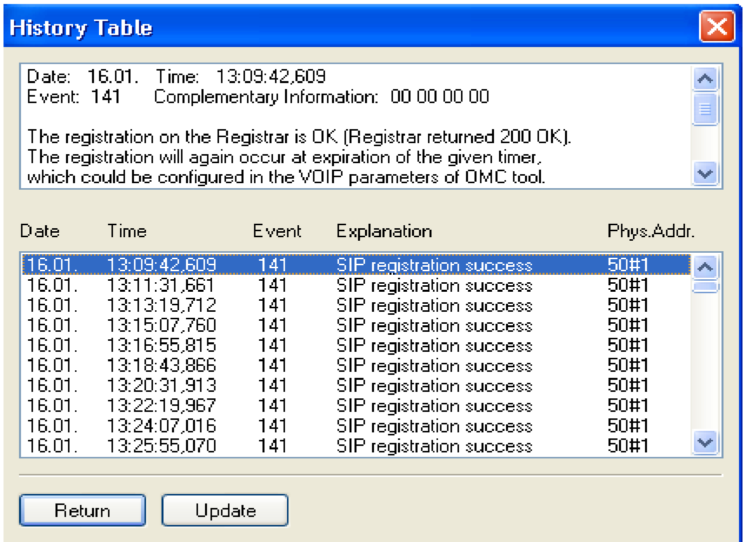
3.3.3 ARS SIP Accounts

Since OXO R920, authentication parameters for SIP gateways have been moved from the OMC section "ARS Gateway Parameters" to the new screen "ARS SIP Accounts". Select the menu Numbering -> Automatic Routing Selection -> SIP Accounts to set up the convenient inputs for SIP gateway index 1.



- ① "Login" : ☞● **GW_Login_example = "thJO******rUk"** (masked partially)
- ② "Password" : ☞● **GW_Password = "*****"** (masked)
- ③ "Registered User." : ☞● **GW_Reg_Username_example = "thJO******rUk"**(masked partially)
- ④ "GW Parms Index" : ☞● **"1"** (index of reference gateway as fixed at Ch. 3.3.2)

At this stage, the IPBX registration (status and logs) can be checked in OMC History&Anomalies → History Table.

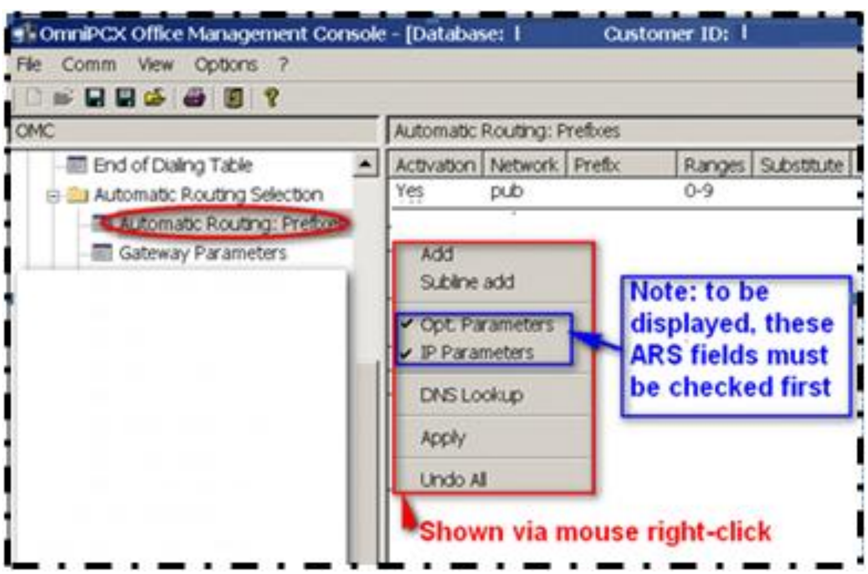


3.3.4 ARS Prefixes

ARS Prefixes are used in the system to build up the routing table of external calls. The initial digits dialed out by a user are looked up in this table as a number prefix. Then, when the digits match an existing prefix of the table, the call is conveyed thru the specific trunk associated to this prefix.

To associate the SIP Trunk to an ARS Prefix, it is necessary to configure this prefix with the IP address of the peering SIP Gateway. **As commented in Ch. 3.3, ARS Prefixes for this SIP Operator must be configured in static IP address mode.**

Go to OMC menu Numbering -> Automatic Routing Selection -> Automatic Routing Prefixes.



As illustrated in the following picture, first insert a route-line for standard calls (i.e. national/ international numbers): use the Add function to create a new line and then, configure the line parameters as indicated.

Automatic Routing: Prefixes (Left part)

Activation	Network	Prefix	Ranges	Subst...	TrGpList	Called(ISVPN/...	User comment	Metering	Calling	Called/PP	Destination	IP
Yes	pub		0-9		1	het	SBC1	Blank	default	default	SIP Gateway	St

Depending on the customer context, you may need to add more route lines for specific number ranges (e.g. short numbers or emergency numbers). Here below is given a more complete example with four Prefixes lines (customized values in area 1 and 2) which permit to route of all the required public phone numbers:

Automatic Routing: Prefixes (left part)

Act...	Network	Prefix	Ranges	Substit...	TrGpList	Called(ISVPN/...	User comment	Counting	Calling	Called/PP	Destination	IP Type
Yes	pub	0	0-9	0	1	het	Line 1	Blank	default	default	SIP Gateway	Dynamic
Yes	emerg				1		Line 2	Blank	default	default		
Yes	pub	3	0-9	3	1		Line 3	Blank	default	pub sh...		
Yes	pub	1	0-9	1	1		Line 4	Blank	default	pub sh...		

- Line 1: standard calls starting with digit 0 (national and international calls)
- Line 2: for external emergency numbers (e.g.112). This line is linked to the system emergency numbers specified in the system flag table "EmergNum" (refer to Technical Bulletin TC80)
- Line 3 and 4: for external short numbers. Depending on the country, the complete list of short numbers will require one or several ARS lines.
    - Line 3: example for France, for short numbers that begin with digit 3 (e.g. 3611, 3900, …)
    - Line 4: example for France, for short numbers that begin with digit 1 (e.g. 11, 118712, …)

In area 2, "Calling" and "Called/PP" fields must be set as shown in the example. In area 3 and 4, values must also be respected:

- "Called (ISVPN/..)": ☞● **ARS_Called_Mode = het**
- "Destination" : ☞● **ARS_Destination = SIP Gateway**

Automatic Routing: Prefixes (right part)

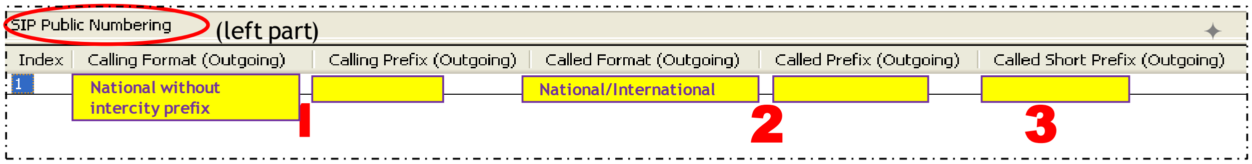
IP Type	IP Address	Hostname	Gateway Alive Protocol	Gateway Alive Ti...	Gateway Bandwidth	Codec/Framing	Gateway Alive Status	Index of Gateway Pa
Dynamic					512 kBit/s	Default	Alive	1
							Alive	1
							Alive	1
							Alive	1

- ① "IP Type": ☞● **ARS_IP_Type = Dynamic** (see note below)
- ② "IP Address": ☞● **ARS_IP_Address = (N/A)** (see note below)
- ③ "GW Alive Protocol": ☞● **ARS_GWalive_Prot = (N/A)** (see note below)
- ④ "GW Alive Timer": ☞● **ARS_GWalive_Timer = (N/A)** (see note below)
- ⑤ "Gateway Bandwidth": to be configured according to the customer IP infra (real bandwidth available for VoIP calls) ☞● **ARS_GW_Bwidth_example = 512 kBit/s**
- ⑥ "Codec/Framing": ☞● **ARS_Codec_Framing = Default**
- ⑦ The Gateway Alive Status is modified when OXO performs a contact with the concerned gateway
- The Index of Gateway field is accessible after the Gateway parameters menu is completed (step 3.5.2)

If GW parameter is set to "DNS SRV" at Ch.3.5.2, then the field ① (IP type) is automatically set to "Dynamic" and the fields②, ③, ④ get disabled.

3.3.5 ARS SIP Public Numbering

The menu Numbering -> Automatic Routing Selection -> SIP Public Numbering is the place to define the format of phone numbers transmitted over the SIP trunk.

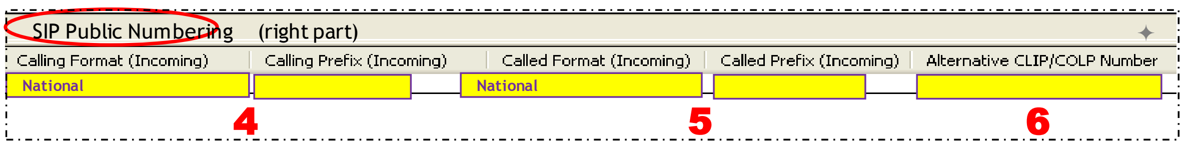


- Configuration for outgoing calls - calling number (From): ①

- "Format": ☞● **SIPnum_Out_Calling_Format = National without intercity prefix**
- "Prefix": ☞● **SIPnum_Out_Calling_Prefix = ""**

- Configuration for outgoing calls - called number (To): ② ③

- "Format": ☞● **SIPnum_Out_Called_Format = National/International**
- "Prefix": ☞● **SIPnum_Out_Called_Prefix = ""**
- "Short Prefix": ☞● **SIPnum_Out_Called_Short_Prefix = ""**



- Configuration for Incoming calls - calling number (From): ④

- "Format": ☞● **SIPnum_Inc_Calling_Format = National**
- "Prefix": ☞● **SIPnum_Inc_Calling_Prefix = ""**

- Configuration for Incoming calls - called number (To): ⑤

- "Format": ☞● **SIPnum_Inc_Called_Format = National**
- "Prefix": ☞● **SIPnum_Inc_Called_Prefix = ""**

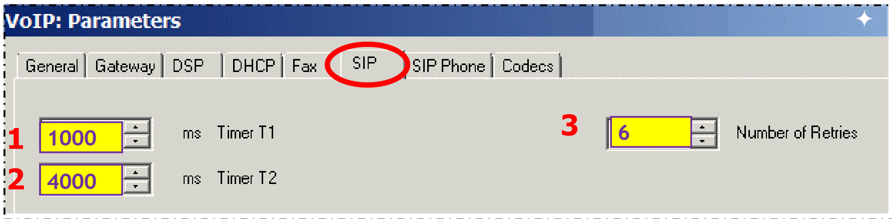
- Configuration for Alternative CLIP/COLP for out-calls - calling number (From): ⑥

- "Alternative CLIP/COLP...":
  ☞● **SIPnum_Alt_CLIP_example = ""**

3.4 VOIP Misc Parameters

3.4.1 SIP Timers

Go to OMC menu Voice Over IP -> VOIP:Parameters/SIP, then Check/ Set values as follows:

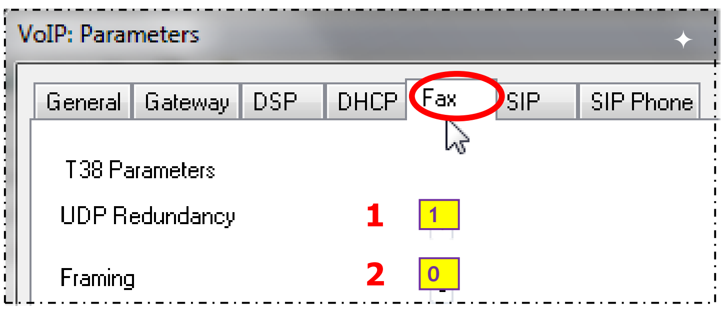


- ① "Timer T1": ☞● **VoIP_SIP_Timer_T1 = "1000"**
- ② "Timer T2": ☞● **VoIP_SIP_Timer_T2 = "4000"**
- ③ "Number of Retries": ☞● **VoIP_SIP_N_Retries = "6"**

3.4.2 Fax (T38)

G711 pass-thru is validated in ARS. Below parameters are just reminded as the default T38 settings of the system.

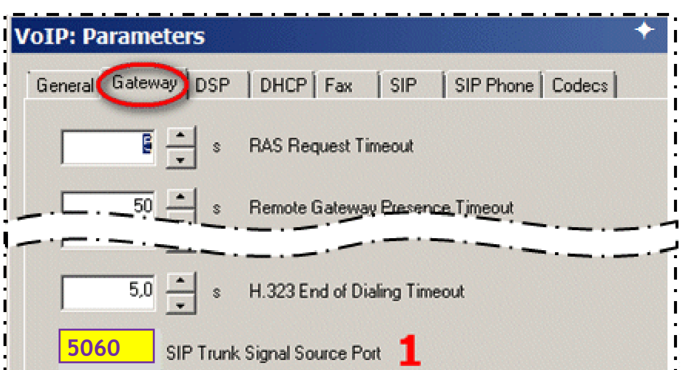
Go to OMC menu Voice Over IP -> VOIP:Parameters/Fax presented below



- ① "UDP Redundancy": ☞● **VoIP_T38_UDP_Redundancy = "1"**
- ② "Framing": ☞● **VoIP_T38_Fax_Framing = "0"**

3.4.3 UDP Source Port for SIP Signaling

Go to OMC menu Voice Over IP -> VOIP:Parameters/Gateway to check/ set up the appropriate input:



- "SIP Trunk Signal Source Port" ①: ☞● **VoIP_SIPSourcePort = "5060"**

3.5 Noteworthy Addresses

Some specific configuration may not be accessible thru a dedicated OMC section/ screen but can be reached via internal flags also referred as “Noteworthy addresses” or "Labels". The access to the system flags is made with the OMC menu System Miscellaneous -> Memory Read/Write.

Noteworthy addresses have a factory default value which is dependent on the installation target selected for OXO software (country dependent).

The system flags detailed hereafter are key for the configuration of the SIP trunk and therefore they need to be checked/ edited with special care. When a flag has been edited, click on the OMC buttons 'Modify' then 'Write' to send the new value to the IPBX (when specified, a further system warm reset may be needed to validate the change).

3.5.1 Debug Labels

Go to System Miscellaneous ->Memory Read/Write > Debug Labels and check/ edit values carefully as detailed below.

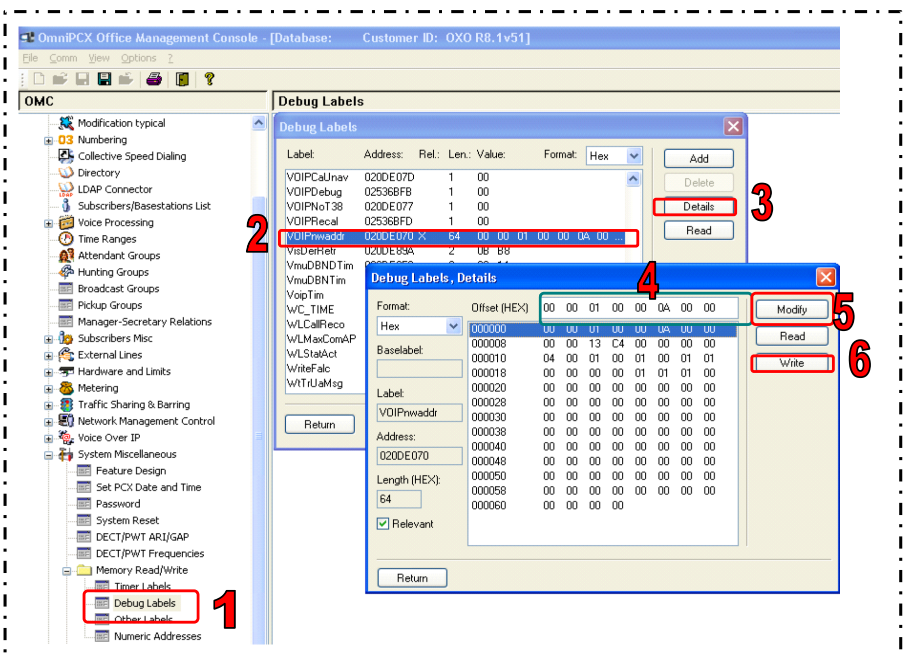
MultAnsRei controls emission of Re-INVITE messages during SDP negotiation

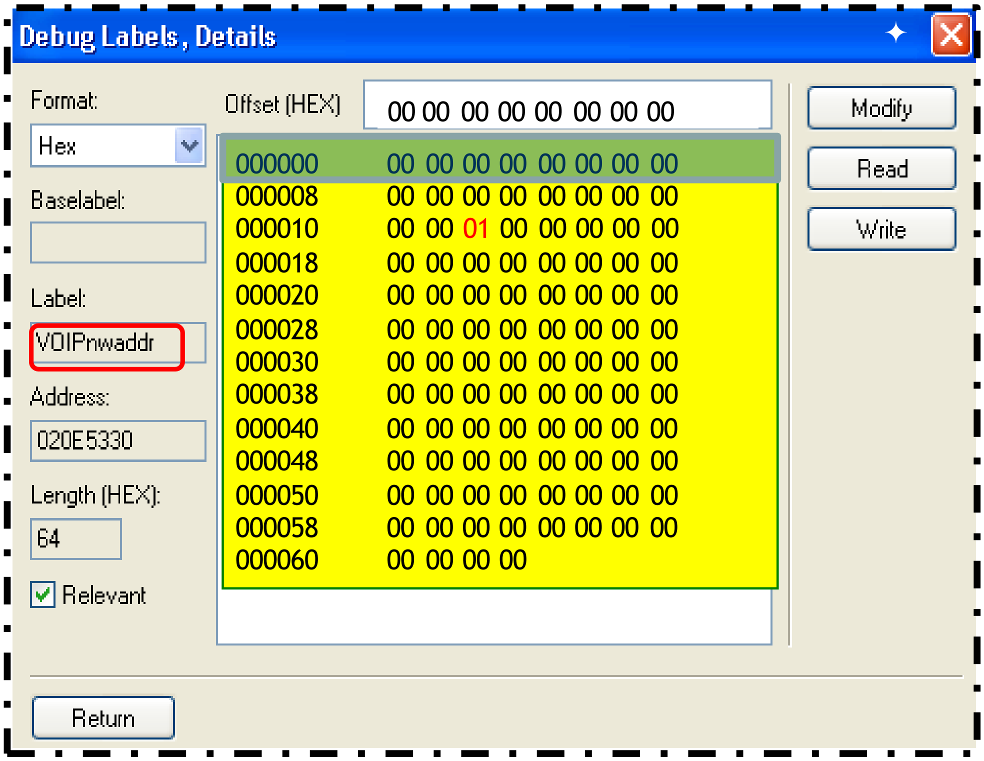
☞● **Flag_MultAnsRei = "01"**

SuprAlerTo controls Ring Back Tone

☞● **Flag_SuprAlerTo = "00"**

VOIPnwaddr is a 64-byte-size table of flags controlling most of the VoIP/SIP parameters outside the regular config screens of OMC. **Pay very special attention to it.**



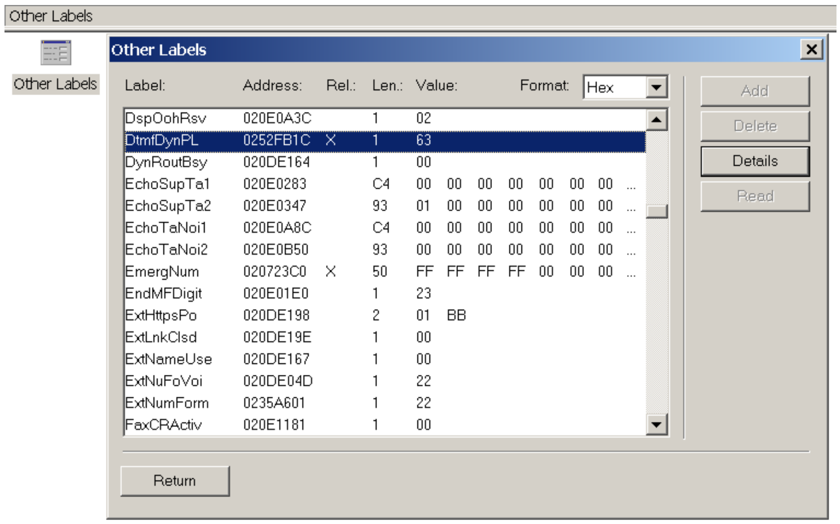


- Line 1 (000:) : ☞● Flag_VOIPnwaddr_Line1 = "00 00 00 00 00 00 00 00"
- Line 2 (008:) : ☞● Flag_VOIPnwaddr_Line2 = "00 00 00 00 00 00 00 00"
- Line 3 (010:) : ☞● Flag_VOIPnwaddr_Line3 = "00 00 **01** 00 00 00 00 00"
- Line 4 (018:) : ☞● Flag_VOIPnwaddr_Line4 = "00 00 00 00 00 00 00 00"
- Line 5 (020:) : ☞● Flag_VOIPnwaddr_Line5 = "00 00 00 00 00 00 00 00"
- Line 6 (028:) : ☞● Flag_VOIPnwaddr_Line6 = "00 00 00 00 00 00 00 00"
- Line 7 (030:) : ☞● Flag_VOIPnwaddr_Line7 = "00 00 00 00 00 00 00 00"
- Line 8 (038:) : ☞● Flag_VOIPnwaddr_Line8 = "00 00 00 00 00 00 00 00"
- Line 9 (040:) : ☞● Flag_VOIPnwaddr_Line9 = "00 00 00 00 00 00 00 00"
- Line 10 (048:) : ☞● Flag_VOIPnwaddr_Line10 = "00 00 00 00 00 00 00 00"

For any modification of this table, a warm reset is needed to validate the change !

3.5.2 Other Labels

Go to System Misc ->Memory Read/Write > Other Labels section and configure labels as shown after.



VipPuNuA controls the global public/ private numbering model. Value must be set to 00.

- **Flag_VipPuNuA = "00"**

ExtNuFoVoi controls usage of the Installation Numbers table. Value must be set to 22.

- **Flag_ExtNuFoVoi = "22"**

DtmfDynPL: Sets the payload value for DTMF

- **Flag_DtmfDynPL = "6A"**

SimulIpAlt controls usage of the local dial tone.

- **Flag_SimulIpAlt = "00"**

PrefCodec/ **PrefFramin** particular codec/ framing preference

- **Flag_PrefCodec = "00 00"**   ● **Flag_PrefFramin = "00"**

FaxPasCd configures codec pass-through preference for fax G711 (between G711a and G711u)

- **Flag_FaxPasCd = "01 FF"**

AlCodLst permits to select one specific default codec list (deprecated since OXO R910)

- **Flag_AlCodLst = (N/A)**

SIPInDspNm controls display of received "CNIP" name when receiving a call

- **Flag_SIPInDspNm = "00"**

SIPOgDspNm controls the "CNIP" name sent over the network for outgoing calls

- **Flag_SIPOgDspNm = "01"**

INVwSDPtrk controls generating INVITEs with/ without SDP for call scenaree with delayed re-routing.

- **Flag_INVwSDPtrk = "00"**

G7222DynPL Sets the payload value for G722-2 codec

- **Flag_G7222DynPL = "75"**

SIPdtmfInB sets alternative DTMF mode for inbound calls

- **Flag_SIPdtmfInB = "00"**

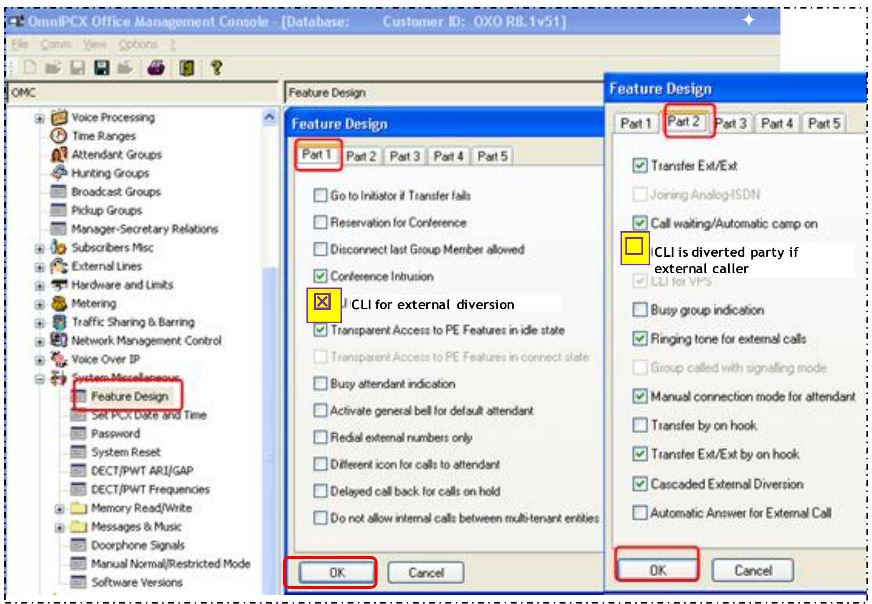
3.6 Miscellaneous Configuration

3.6.1 Calling Identity when External Call Forward

This configuration permits to define the calling CLIP that is sent by OXO to the "forwarded-to" destination (case of an incoming external call coming to an internal user who has an external diversion engaged). The control can be made globally for the PBX or extension by extension.

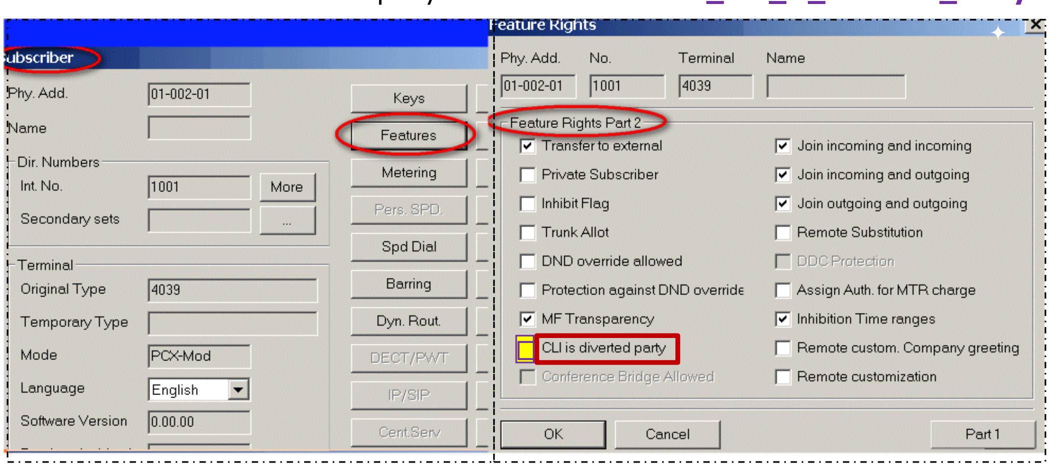
From the menu System Misc -> Feature Design, tab Part 1 & tab Part 2; check the parameters:

- "CLI for external diversion": ☞● **Misc_CLI_Ext_Diversion = True**
- "CLI is diverted party if ext..": ☞● **Misc_CLI_is_Diverted_Party = False**



For extensions taken from the menu **Subscribers/Basestation List**, check with the **Details** button the similar CLI parameter:

- "CLI is diverted party": ☞● **Misc_CLI_is_Diverted_Party = False**



4 SIP trunk Configuration Abstract

This chapter gathers the rough value of OXO parameters (three following tables).

Table 1

CONFIG OXO R920	VALUE	REMARK
LAN_IP_Infra		
LAN_Main_CPU_IPadd_example	"192.168.137.246"	Value given as example
LAN_Def_Router_IPadd_example	"192.168.137.254"	Value given as example
LAN_Subnet_Mask_example	"255.255.255.0"	Value given as example
System_Numbering_Plan		
NP_Instal_Number_example	"9****3430"	Value given as example
NP_International_Prefix	"00"	
NP_International_Code	"33"	
NP_Recall_Prefix	"0"	
NP_Intercity_Prefix	"0"	
NP_Intercity_Code_example	""	
NP_System_Alt_CLIP_example	""	
NP_DDI_Range_example	"3431 3435"	Value given as example
VoIP_General		
VoIP_Protocol	SIP	
VoIP_Trunk_Channels_example	"4"	Value given as example
VoIP_RTP_Direct	False	
VoIP_Trunk_Codec_Passthru	False	
VoIP_Phone_Codec_Passthru	False	
VoIP_G711_MOH	False	
VoIP_IP_QoS_example	10111000 DIFFSERV_PHB_EF	Value given as example
VoIP_DSP_Echo_Cancel	True	
VoIP_DSP_VAD	False	
VoIP_T38_UDP_Redundancy	"1"	
VoIP_T38_Fax_Framing	"0"	
VoIP_SIP_Timer_T1	"1000"	
VoIP_SIP_Timer_T2	"4000"	
VoIP_SIP_N_Retries	"6"	
VoIP_SIPSourcePort	"5060"	
VoIP_Def_CodecFraming	"20 ms"	
VoIP_Def_CodecList	G711.µ	
Trunk_Access		
Access_is_Public	True	
Access_Alt_CLIP_example	""	
Access_Channels_example	"4"	Value given as example
System_Misc		
Misc_CLI_Ext_Diversion	True	
Misc_CLI_is_Diverted_Party	False	

Table 2

CONFIG OXO R920	VALUE	REMARK
ARS_Prefixes		
ARS_Called_Mode	het	
ARS_Destination	SIP Gateway	
ARS_IP_Type	Dynamic	
ARS_IP_Address	(N/A)	
ARS_GWalive_Timer	(N/A)	
ARS_GWalive_Prot	(N/A)	
ARS_GW_Bwidth_example	512 kBit/s	Value given as example
ARS_Codec_Framing	Default	
ARS_GW_Parms		
GW_Login_example	"thJO******rUk"	Value given as example
GW_Password	"*****"	
GW_Target_Domain	"sip9.voip-centrex.net"	
GW_Realm	""	
GW_RFC3325	Yes	
GW_DNS_Mode	DNSSRV	
GW_Remote_SIP_Port	"5060"	
GW_Fax_Mode	G711	
GW_Local_Domain_Name	""	
GW_Prim_DNS	"83.167.37.51"	
GW_Sec_DNS	"8.8.8.8"	
GW_Outb_Proxy	"sip9.voip-centrex.net"	
GW_Reg_Requested	Yes	
GW_Reg_PubDDI	No	
GW_Reg_Username_example	"thJO******rUk"	Value given as example
GW_Reg_IP_Address	(N/A)	
GW_Reg_Port	"(Dynamic)"	
GW_Reg_Name	"sip9.voip-centrex.net"	
GW_Reg_Expire_Time	"60"	
ARS_SIP_Public_Numbering_Plan		
SIPnum_Out_Calling_Format	National without intercity prefix	
SIPnum_Out_Calling_Prefix	""	
SIPnum_Out_Called_Format	National/International	
SIPnum_Out_Called_Prefix	""	
SIPnum_Out_Called_Short_Prefix	""	
SIPnum_Inc_Calling_Format	National	
SIPnum_Inc_Calling_Prefix	""	
SIPnum_Inc_Called_Format	National	
SIPnum_Inc_Called_Prefix	""	
SIPnum_Alt_CLIP_example	""	

Table 3

CONFIG OXO R920	VALUE	REMARK
System_Flags		
Flag_VOIPnwaddr_Line1	"00 00 00 00 00 00 00 00"	
Flag_VOIPnwaddr_Line2	"00 00 00 00 00 00 00 00"	
Flag_VOIPnwaddr_Line3	"00 00 **01** 00 00 00 00 00"	
Flag_VOIPnwaddr_Line4	"00 00 00 00 00 00 00 00"	
Flag_VOIPnwaddr_Line5	"00 00 00 00 00 00 00 00"	
Flag_VOIPnwaddr_Line6	"00 00 00 00 00 00 00 00"	
Flag_VOIPnwaddr_Line7	"00 00 00 00 00 00 00 00"	
Flag_VOIPnwaddr_Line8	"00 00 00 00 00 00 00 00"	
Flag_VOIPnwaddr_Line9	"00 00 00 00 00 00 00 00"	
Flag_VOIPnwaddr_Line10	"00 00 00 00 00 00 00 00"	
Flag_VipPuNuA	"00"	
Flag_ExtNuFoVoi	"22"	
Flag_DtmfDynPL	"6A"	
Flag_MultAnsRei	"01"	
Flag_SimulIpAlt	"00"	
Flag_PrefCodec	"00 00"	
Flag_PrefFramin	"00"	
Flag_FaxPasCd	"01 FF"	
Flag_SIPInDspNm	"00"	
Flag_SIPOgDspNm	"01"	
Flag_INVwSDPtrk	"00"	
Flag_SIPdtmfInB	"00"	
Flag_G7222DynPL	"75"	
Flag_AlCodLst	(N/A)	
Flag_SuprAlerTo	"00"	

Follow us on Facebook and Twitter

Stay tuned on our Facebook and Twitter channels where we inform you about:

- New software releases
- New technical communications
- AAPP InterWorking Reports
- Newsletter
- Etc.

twitter.com/ALUE_Care

facebook.com/ALECustomerCare

Submitting a Service Request

Please connect to our eService Request application.

Before submitting a Service Request, make sure that:

- In case a Third-Party application is involved, that application has been certified via the AAPP
- You have read through the Release Notes which lists new features available, system requirements, restrictions etc. available in the Technical Documentation Library
- You have read through the Troubleshooting Guides and Technical Bulletins relative to this subject available in the Technical Documentation Library
- You have read through the self-service information on commonly asked support questions, known issues and workarounds available in the Technical Knowledge Center

- END OF DOCUMENT -