



2N<sup>®</sup>

# VoiceBlue Lite



## 2N<sup>®</sup> VoiceBlue Lite & Alcatel OXO

connected via SIP trunk

Quick guide

Version 1.00

[www.2n.cz](http://www.2n.cz)

## 2N® VoiceBlue Lite has these parameters:

- IP address 192.168.92.200
- Incoming port: 5060
- Firmware version: 2.07.35i39

## Alcatel OXO parameters:

- IP address 192.168.92.246
- Incoming port: 5060
- version 7.7.1

## SIP TRUNK INTERCONNECTION

- 1) For the setting of the trunk between the VoiceBlue Lite and your PBX you need to configure SIP proxy (GSM→IP) for GSM incoming calls. The setting is in the VoIP menu. SIP proxy (IP→GSM) is designed for secure communication just with traffic from your PBX. You can specify the IP address and port which will accept SIP packets from.

In case you leave there 0.0.0.0 it will be open for all traffic.

The screenshot shows a configuration interface for SIP settings. It includes fields for IP addresses, checkboxes for 'Use default port', and input fields for ports. Two callouts provide additional context: one points to the IP address field for 'SIP proxy (IP->GSM)' and another points to the port field for the same setting.

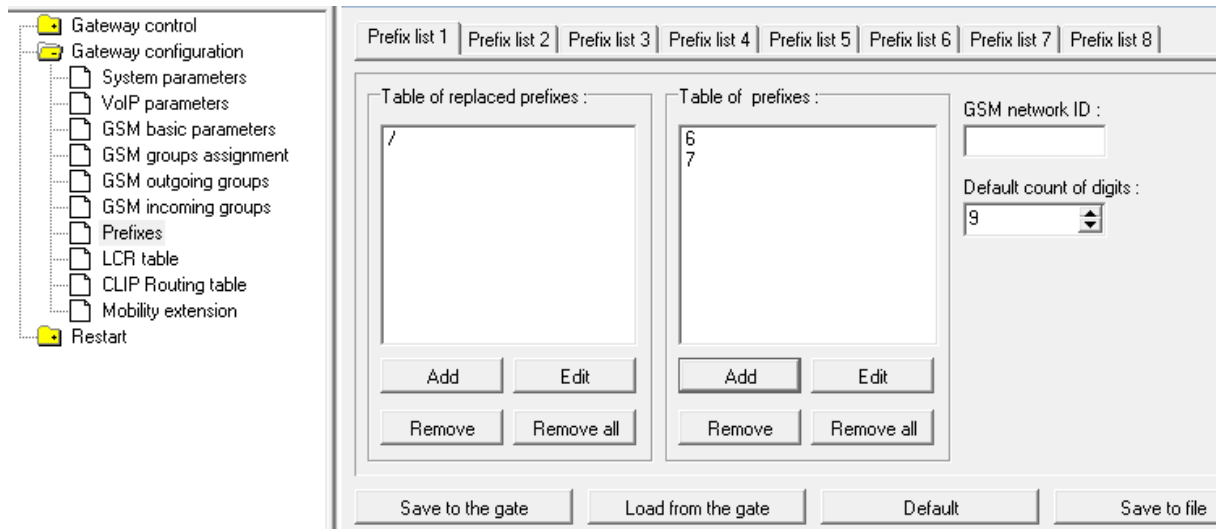
Setting	IP Address	Use default port	Port
SIP proxy ( IP->GSM ) :	192.168.92.246	<input checked="" type="checkbox"/>	5060
SIP proxy ( GSM-> IP)	192.168.92.246	<input checked="" type="checkbox"/>	5060
SIP registrar :	0.0.0.0	<input checked="" type="checkbox"/>	5060
NAT firewall :	0.0.0.0		
STUN server :	0.0.0.0	<input checked="" type="checkbox"/>	3478
Next STUN server request : ( 60 - 6553s)	600		[s]

Tones generated to VoIP :

Dial tone to VoIP : Transfer from GSI

## 2) Configuration of the LCR (Least Cost Routing)

The GSM operator has e.g. in our country prefix 6 and 7 with a nine digit the length number.  
The setting is below.



- 3) You need to create LCR rule for defined prefixes. The GSM group says thru with outgoing group the call will follow and in the GSM group assignment you can define, which SIM card belongs to which GSM outgoing group.

LCR table			
Prefix List	Valid from/to	Outgoing destination	Call duration limit
1/	0:00/24:00	1	0
2/	00:00/24:00	2	0

Add

Edit

Remove

Remove all

Load from the gate

Save to the gate

Save to file

Default

The screenshot shows the 'Gateway control' software interface. The main window has a menu bar with 'File', 'Gateway', 'Gateway control', 'Settings', and 'Help'. Below the menu bar is a toolbar with various icons. The left sidebar shows a tree view under 'Topics' with 'Alphabetical glossary' selected. The tree view includes 'Gateway control', 'Gateway configuration', 'System parameters', 'VoIP parameters', 'GSM basic parameters', 'GSM groups assignment', 'GSM outgoing groups', 'GSM incoming groups', 'Prefixes', 'LCR table', 'CLIP Routing table', 'Mobility extension', and 'Restart'. The main content area displays the 'Groups assignment' configuration window. This window has a table with columns 'Module', 'Outgoing', and 'Incoming'. The table contains four rows for modules 0, 1, 2, and 3. Below the table are four buttons: 'Save to the gate', 'Load from the gate', 'Default', and 'Save to file'.

Module :	Outgoing :	Incoming :
0. module	1. Group	1. Group
1. module	1. Group	1. Group
2. module	2. Group	1. Group
3. module	2. Group	1. Group

Buttons: Save to the gate, Load from the gate, Default, Save to file

#### 4) Configuration of GSM outgoing groups:

You are able to set up different setting for each GSM group (CLIR, free minutes, Virtual ring tone, roaming and others). In case you don't have a Ring back tone, set up Delay for ALERTING to option 4.

GSM outgoing groups

1 GSM group | 2 GSM group | 3 GSM group | 4 GSM group

Roaming enabled for network code:  Delay for CONNECT [s]:

CLIR:  Delay for ALERTING [s]: (0 = off)

Max. number of called minutes: (0 = off)  Day of deleting stats in group:

SMS messages number: (0 = off)  Minimal ring duration to send "SMS at no answer" [s]: (0 = off)

Day of deleting stats: (0 = off)  Text of "SMS at no answer":

Minimal length of call after connect [s]:  Text of SMS for all calls (number = %N):

Precision of counting length of call:  CLIP to GSM separator:

CLIP to GSM modification:

Use CLIP to GSM from INVITE field:

For proper functionality "Clip to GSM separator" has to be set

Save to the gate | Load from the gate | Default | Save to file

#### 5) Incoming calls

For incoming calls you can define 2 groups with the different behavior and assign them to the GSM modules. The settings are similar with GSM groups assignment for outgoing calls.

File Gateway Gateway control Settings Help

Topics | Alphabetical glossary

- Gateway control
- Gateway configuration
  - System parameters
  - VoIP parameters
  - GSM basic parameters
  - GSM groups assignment
  - GSM outgoing groups
  - GSM incoming groups
  - Prefixes
  - LCR table
  - CLIP Routing table
  - Mobility extension
- Restart

Groups assignment:

Module:	Outgoing:	Incoming:
0. module	<input type="text" value="1. Group"/>	<input type="text" value="1. Group"/>
1. module	<input type="text" value="1. Group"/>	<input type="text" value="1. Group"/>
2. module	<input type="text" value="2. Group"/>	<input type="text" value="1. Group"/>
3. module	<input type="text" value="2. Group"/>	<input type="text" value="1. Group"/>

Save to the gate | Load from the gate

Default | Save to file

In GSM incoming groups you can define the behavior for each GSM incoming group. Choose the mode to Reject, Ignore, Accept incoming calls or Callback.

**GSM incoming groups**

1 GSM group | 2 GSM group | 3 GSM group | 4 GSM group

Mode :  List of called numbers :  
(Call number by %A, %G95..8 or none or answer and wait for DTMF)

Min. digits in DTMF :

Max. digits in DTMF :

Timeout for entering DTMF digit [s] :

Day of deleting GSM inc. group stastics : (0 = off)

Prefix before DISA preselection :

CLIP :

CLIP to VoIP separator:  (empty = off)

CLIP to VoIP modification :  **For proper functionality "Clip to VoIP separator" has to be set**

Time to keep CLIP in table [hours] :   Off

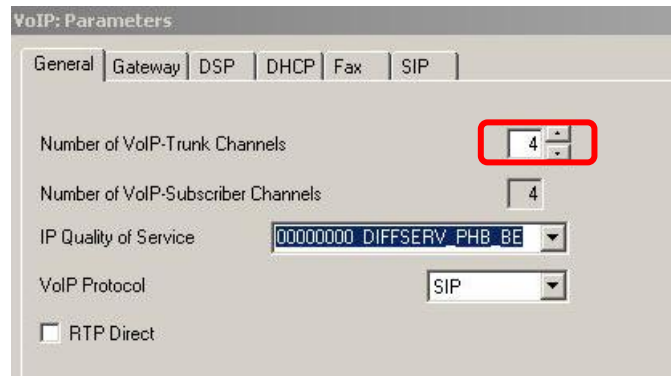
Add record only for unconnected call

Delete record for connected answer

You can define the list of called numbers which will be automatically dialed after DTMF dialing timeout if the customer don't press any button till the specified time. From the configuration, you can see 10 seconds for DTMF dialing and after that the call will be routed to the extension 101 to your PBX (if you set up SIP proxy (GSM->IP) in VoIP parameters).

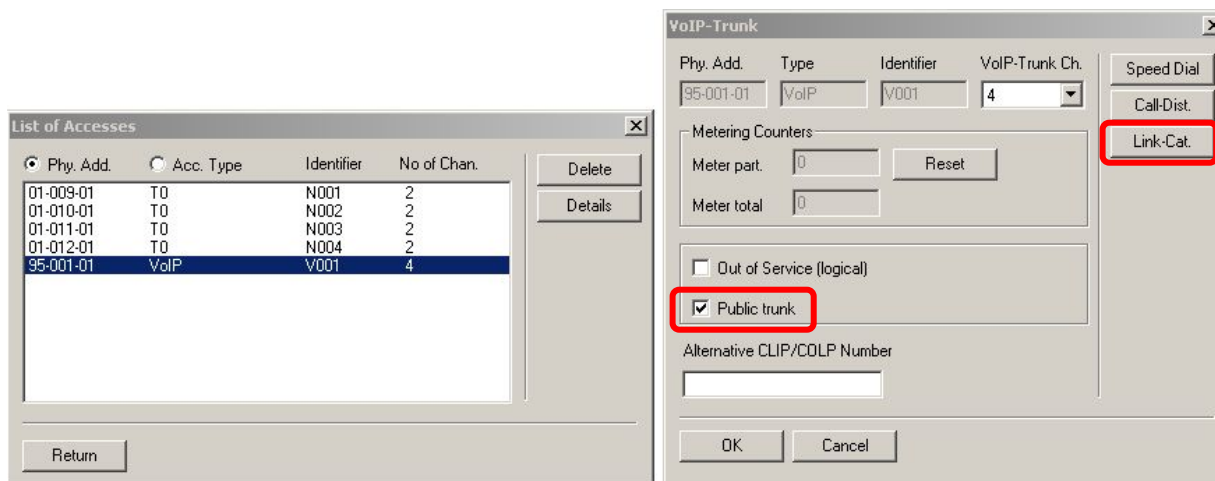
# Alcatel OXO configuration

Setup a count of VoIP-Trunk channels for VoIP trunk to OfficeRoute



The 'VoIP: Parameters' window shows the 'General' tab. The 'Number of VoIP-Trunk Channels' is set to 4, highlighted with a red box. Other settings include 'Number of VoIP-Subscriber Channels' set to 4, 'IP Quality of Service' set to '00000000 DIFFSERV\_PHB\_BE', and 'VoIP Protocol' set to 'SIP'. The 'RTP Direct' checkbox is unchecked.

Choose Trunk group and check Public trunk checkbox. Change Link Category settings

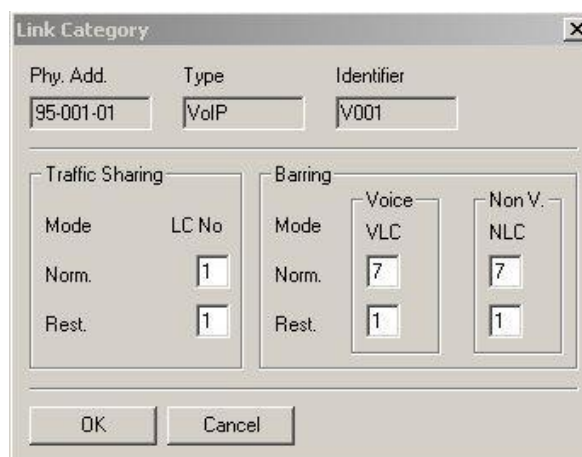


The 'List of Accesses' window shows a table with the following data:

Phy. Add.	Acc. Type	Identifier	No of Chan.
01-009-01	TO	N001	2
01-010-01	TO	N002	2
01-011-01	TO	N003	2
01-012-01	TO	N004	2
95-001-01	VoIP	V001	4

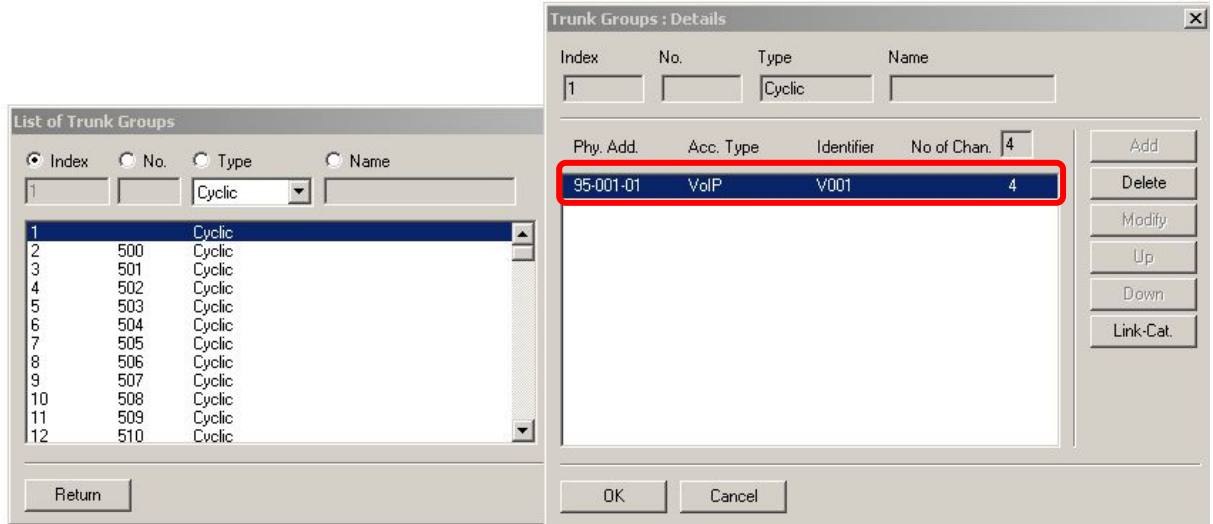
The 'VoIP-Trunk' window shows the configuration for the selected trunk. The 'Public trunk' checkbox is checked, highlighted with a red box. The 'Link-Cat.' button is also highlighted with a red box.

At Link Category menu setup all necessary parametres

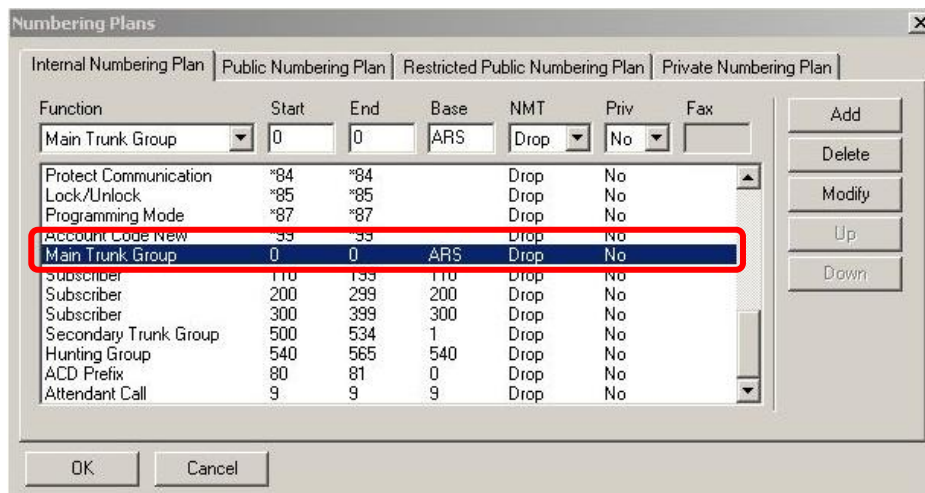


The 'Link Category' window shows the configuration for the selected trunk. The 'Phy. Add.' is '95-001-01', 'Type' is 'VoIP', and 'Identifier' is 'V001'. The 'Traffic Sharing' section has 'Mode' set to 'Norm.' and 'LC No' set to 1. The 'Barring' section has 'Mode' set to 'Norm.' and 'VLC' set to 7, and 'Non V.' set to 7. The 'Rest.' section has 'VLC' set to 1 and 'NLC' set to 1.

## Assign Trunk Groups



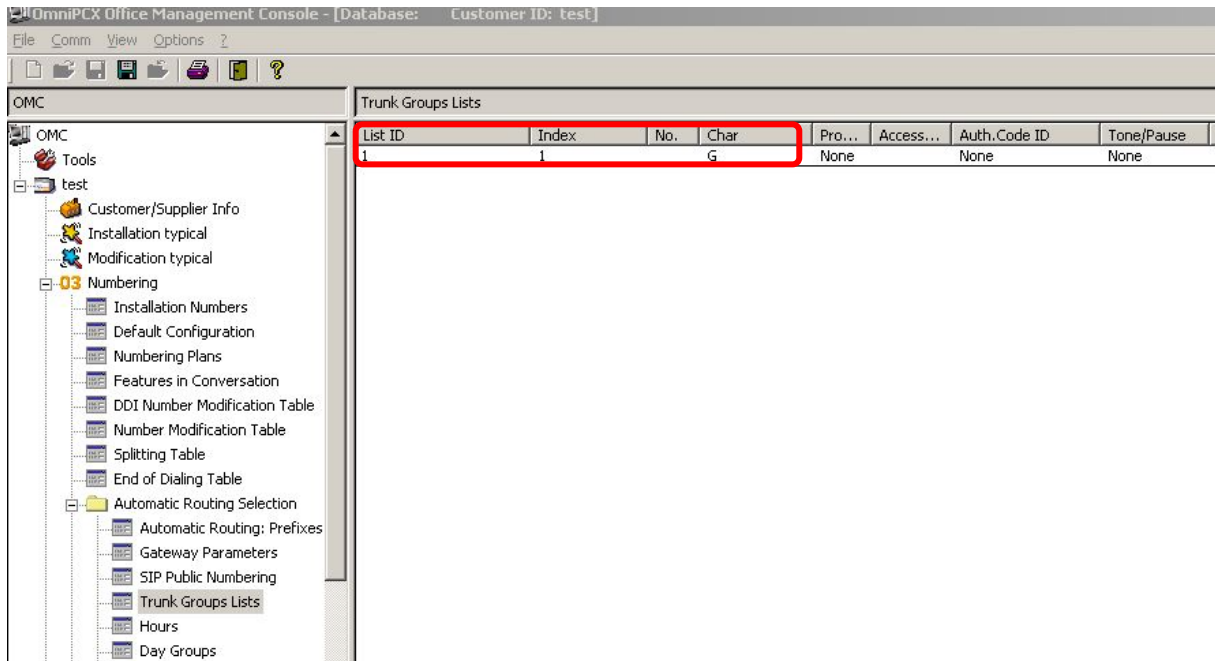
Setup Numbering plans table. As Base settings choose ARS



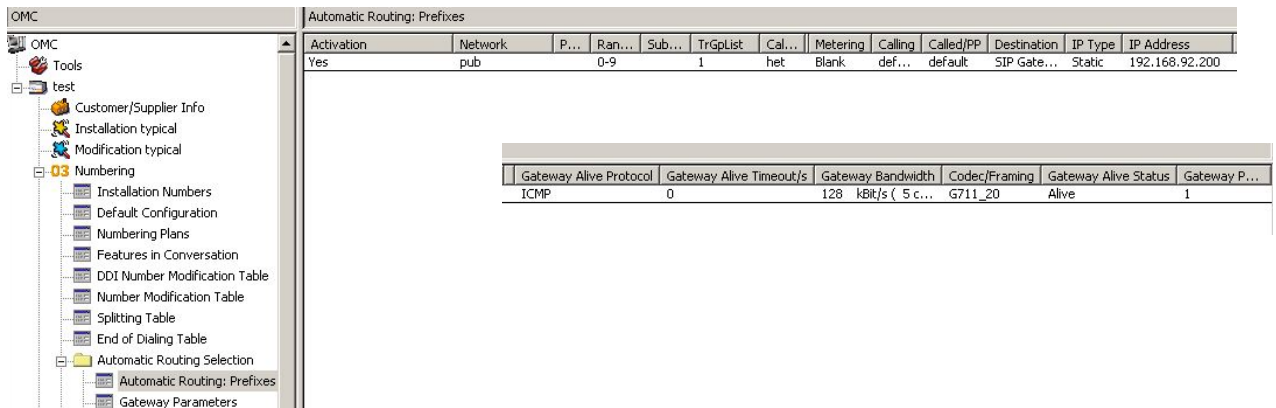
Check Public numbering plan for incoming calls.



At Trunk Groups List assign List ID with Index at menu **Numbering** → **Automatic Routing Selection** → **Trunk Groups List**



Setup IP address of OfficeRoute, codecs, bandwidth and GW keepalive timeout for VoIP trunk at menu **Numbering** → **Automatic Routing Selection** → **Automatic Routing: Prefixes**



At menu **Numbering** → **Automatic Routing Selection** → **Gateway Parametres** setup listening port 5060. If you want to use OfficeRoute with different listening port (for example 5065), just setup the number of the port here.

Gateway Parameters							
Index	Login	Password	Domain Name	Realm	RFC 3325	Re...	SIP Numbers ...
1			192.168.92.200		Yes	5060	1



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