



2N[®]

StarGate BlueTower



2N[®] StarGate/BlueTower & Alcatel OXO

connected via SIP trunk

Quick guide

Version 1.00

www.2n.cz

2N® StarGate has these parameters:

- IP address 192.168.92.200
- Incoming port: 5060
- Firmware version: 2.30.03fxx

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 StarGate and your PBX you need to configure SIP proxy (GSM→IP) for calls coming from GSM. You will do it in VoIP parameters. SIP proxy (IP→GSM) is designed for secure communication just with traffic from your PBX. You can specify the IP address and port from which the gateway will accept SIP packets from.

In case you leave IP→GSM 0.0.0.0 the gateway will accept SIP traffic from all IPs on defined port.

The screenshot shows a configuration window for SIP parameters. It is divided into several sections:

- Voice parameters :** Includes fields for First RTP port (8000) and Last RTP port (8998). A red note states: "1024 and 2nd port must be number of used VoIP".
- Codecs settings :** Includes a section for "Number of blocks" with dropdowns for G711 (4), G723 (1), and G729 (2). There are also checkboxes for VAD.
- IP address and port settings:** Includes fields for SIP proxy (IP→GSM) (192.168.92.246) and SIP proxy (GSM→IP) (192.168.92.146). It also has checkboxes for "Use default port" and corresponding port number fields (5060).
- Other parameters:** Includes fields for SIP registrar (0.0.0.0), NAT firewall (0.0.0.0), VoIP card / MGCP gateway (0.0.0.0), STUN server (0.0.0.0), and Next STUN server request (0 [s]).
- Tones generated to VoIP :** Includes a dropdown for "Dial tone to VoIP" set to "From GSM".

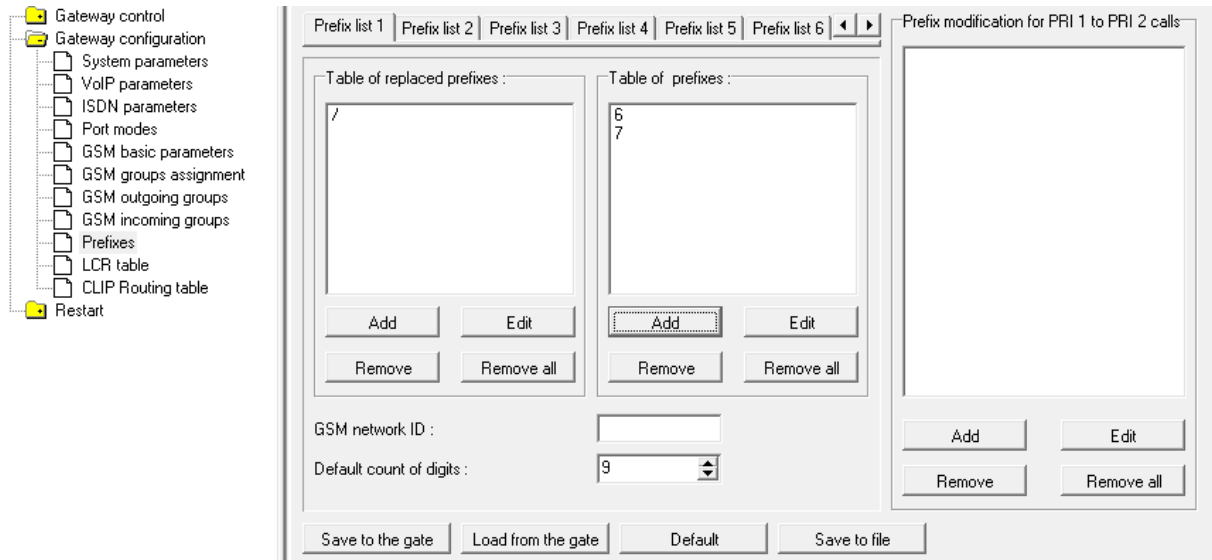
Two callout bubbles are present:

- One pointing to the SIP proxy (IP→GSM) field, containing the text: "The IP address to which the traffic is send when comming from GSM".
- Another pointing to the SIP proxy (IP→GSM) field and its port field, containing the text: "The IP address and port from which the gateway will accept traffic on VoIP interface".

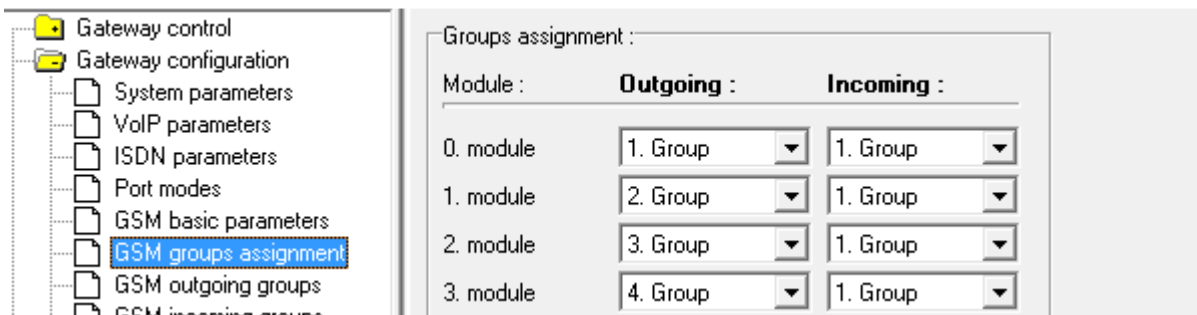
At the bottom, there are buttons for "Save to the gate", "Load from the gate", "Default", and "Save to file".

2) Configuration of the LCR (Least Cost Routing)

The GSM operator uses for instance prefix 6 and 7 with a number length of nine digits. The number should be dialed to GSM in national format. The configuration is as follows.

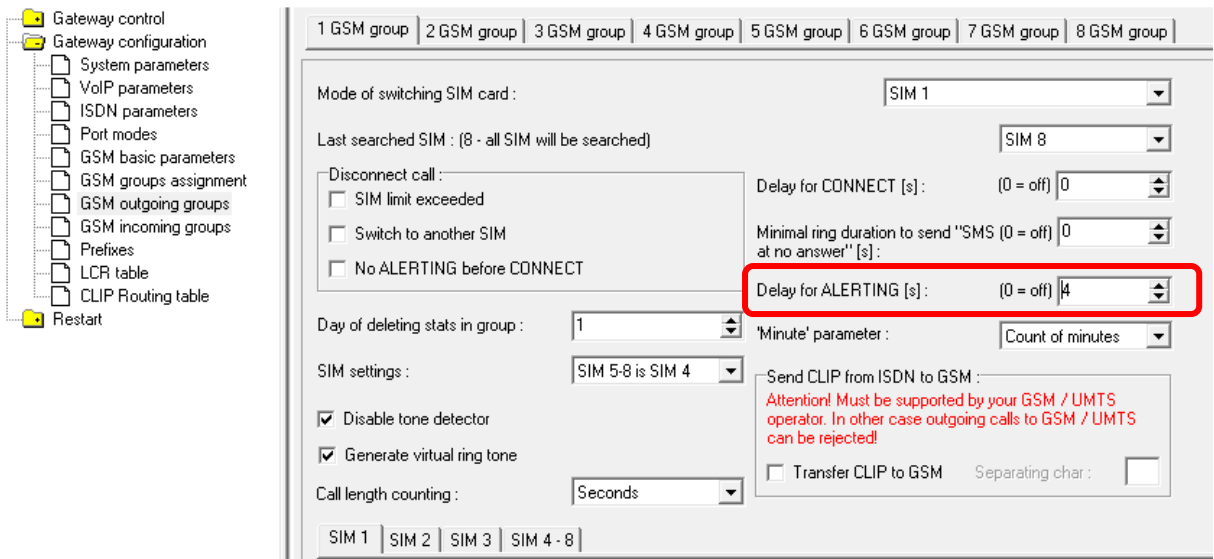


3) You need to create LCR rule for defined prefixes. The GSM group defines thru which outgoing group the call will be handled. In the GSM group assignment you can define, which SIM card bellows to which GSM outgoing group.



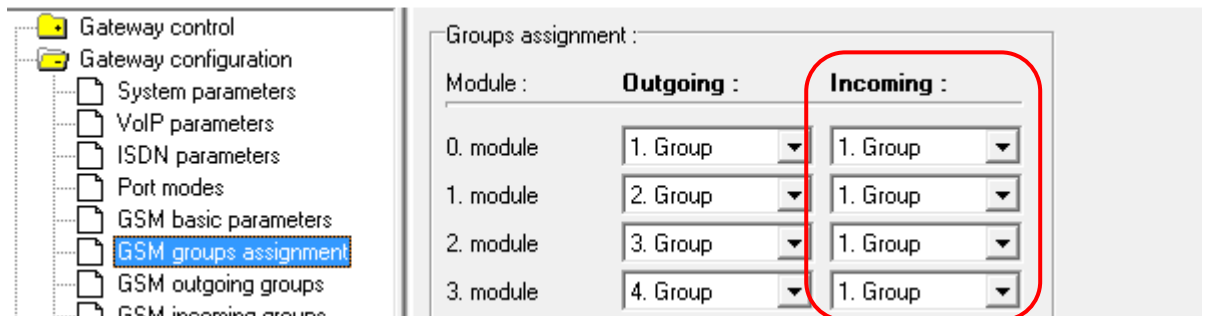
4) Configuration of GSM outgoing groups:

You can use different setting for different GSM group (CLIR, free minutes, Virtual ring tone, roaming and others). Set Delay for ALERTING to option 4 in case you don't have a Ring back tone.



5) Incoming calls

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



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

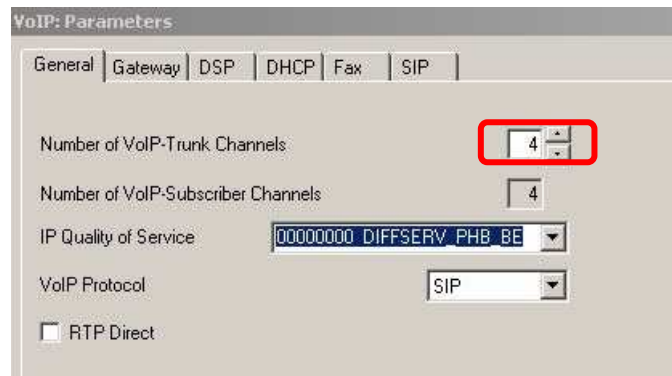
The screenshot displays the configuration interface for GSM incoming groups. On the left, a tree view shows the navigation structure, with 'GSM incoming groups' highlighted. The main configuration area is titled '1 GSM group' and includes the following settings:

- Mode:** A dropdown menu set to 'Accept incoming calls + dialtone'.
- List of called numbers:** A text area containing '101', with 'Add', 'Remove', and 'Remove all' buttons below it.
- (Call number by %A, %G95.8 or none or answer and wait for DTMF)**
- Min. digits in DTMF:** A spinner box set to 3.
- Max. digits in DTMF:** A spinner box set to 3.
- Timeout for entering DTMF digit [s]:** A spinner box set to 10.
- Day of deleting GSM inc. group stastics: (0=off):** A spinner box set to 1.
- Prefix before DISA preselection:** An empty text field.
- CLIP:** An empty text field, with a note: '(+ removed automatically; * removes one digit)'. There is also a '+' symbol next to the field.
- Looping of voice message [min]: (0 = off):** A spinner box set to 0.
- Time to keep CLIP in table [hours]: (0 = off):** A spinner box set to 0.
- CDN recognition in CLIP (Separating char):** An empty text field.

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

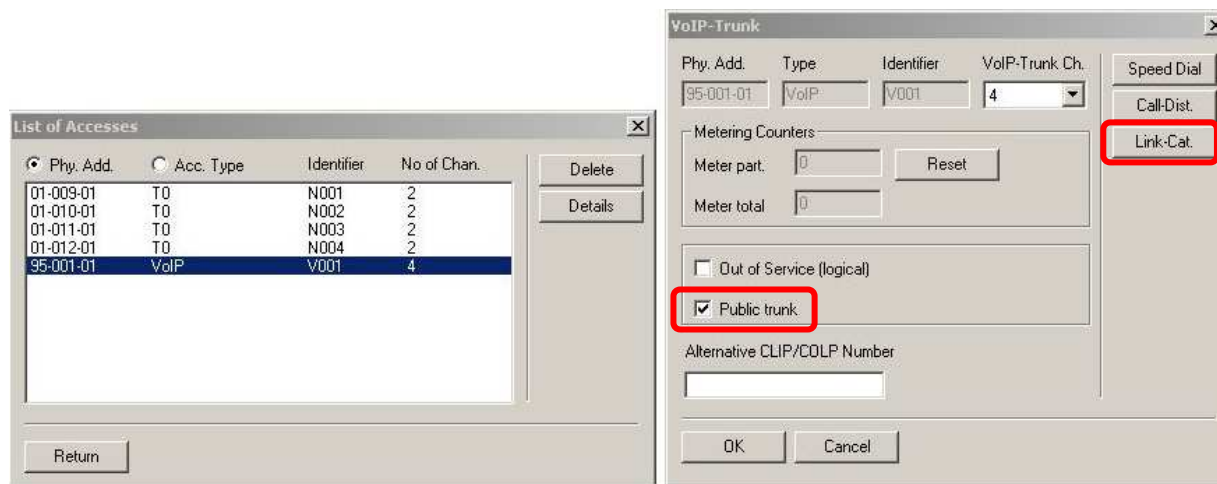
Alcatel OXO configuration

First step is to set a count of VoIP-Trunk channels for VoIP trunk to 2N® StarGate/BlueTower



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 tick Public trunk checkbox. Change Link Category settings

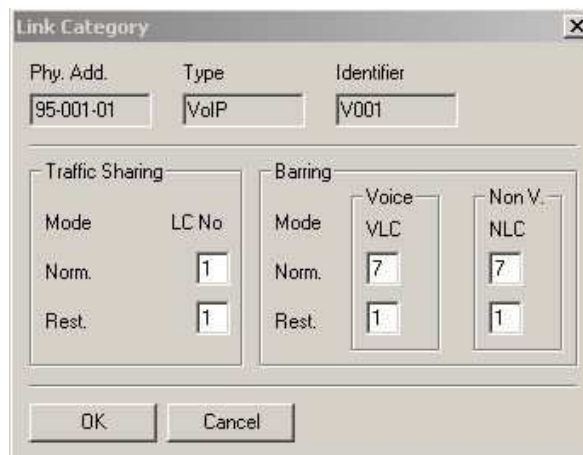


The 'List of Accesses' window shows a table with the following data:

Phy. Add.	Acc. Type	Identifier	No of Chan.
01-009-01	T0	N001	2
01-010-01	T0	N002	2
01-011-01	T0	N003	2
01-012-01	T0	N004	2
95-001-01	VoIP	V001	4

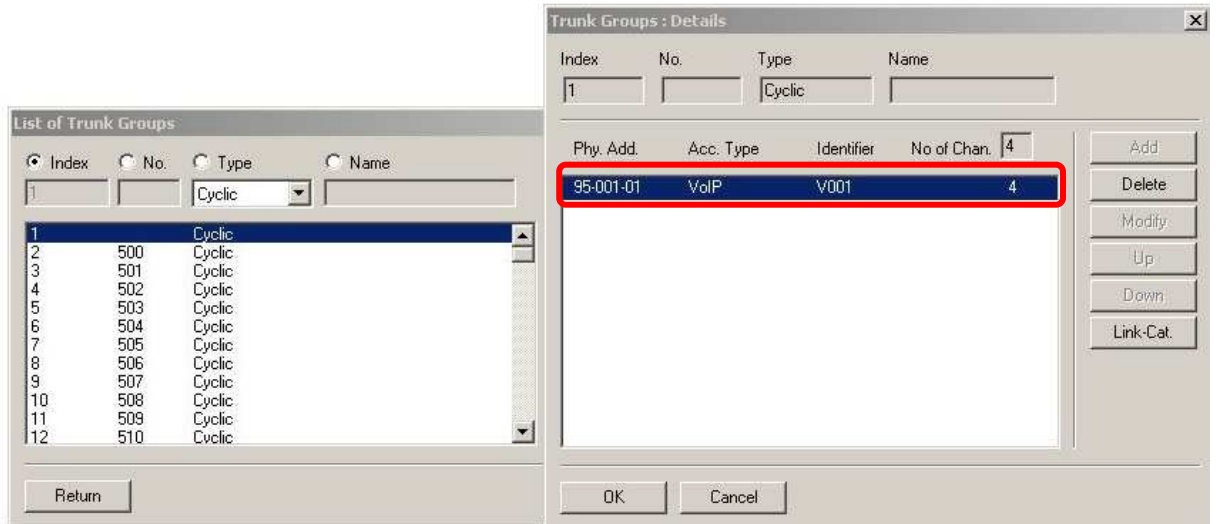
The 'VoIP-Trunk' window shows the configuration for the selected trunk. The 'Public trunk' checkbox is checked, highlighted with a red box. The 'Link-Cat.' button is also highlighted with a red box.

At Link Category menu set up all necessary parameters. Barring is intended for limitation of called numbers (7 is link to a table which allows calls to be made to all possible numbers)

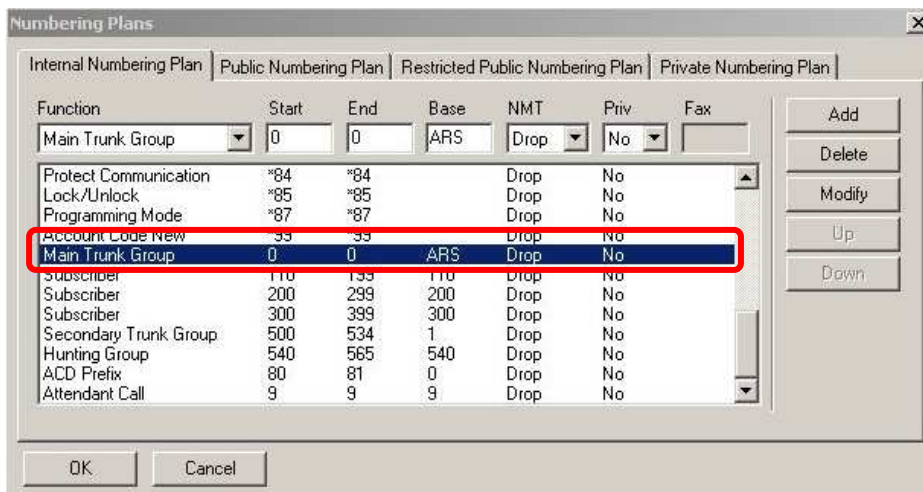


The 'Link Category' window shows the configuration for the selected trunk. The 'Phy. Add.' is 95-001-01, 'Type' is VoIP, and 'Identifier' is V001. The 'Traffic Sharing' section has 'Mode' set to 'Norm.' and 'LC No.' set to 1. The 'Barring' section has 'Mode' set to 'VLC' and 'Non V.' set to 'NLC', with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

Assign Trunk Groups

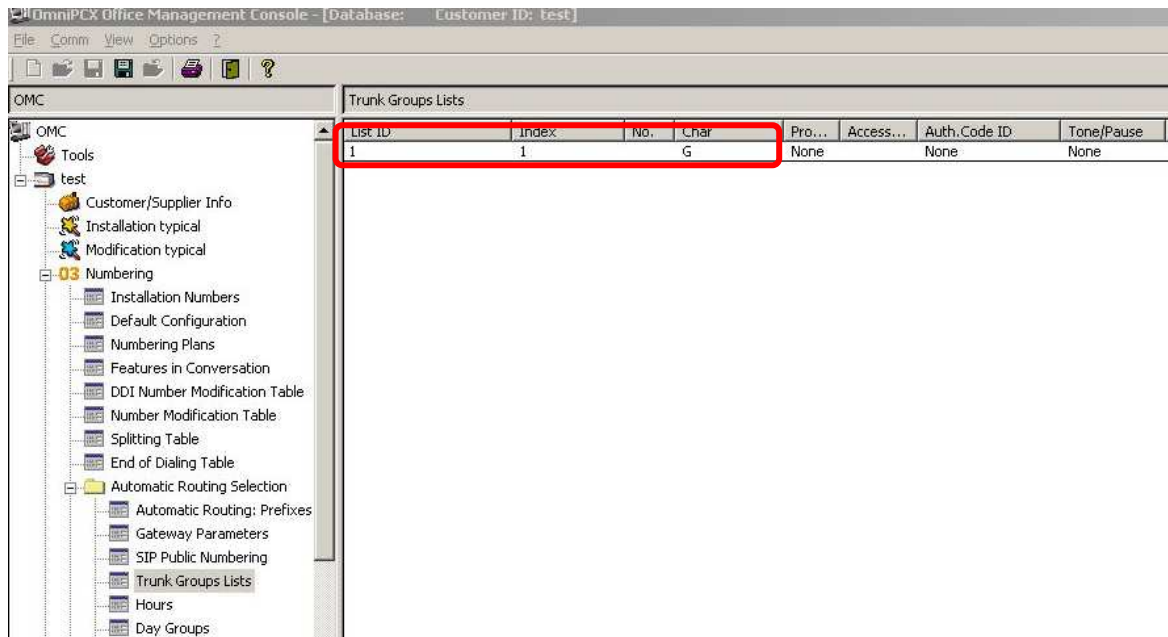


Set Numbering plans table. Choose ARS (automatic routing system) alias LCR for Base settings. Start and End dedicate range of prefixes for outgoing calls from PBX. This prefix is stripped later on.

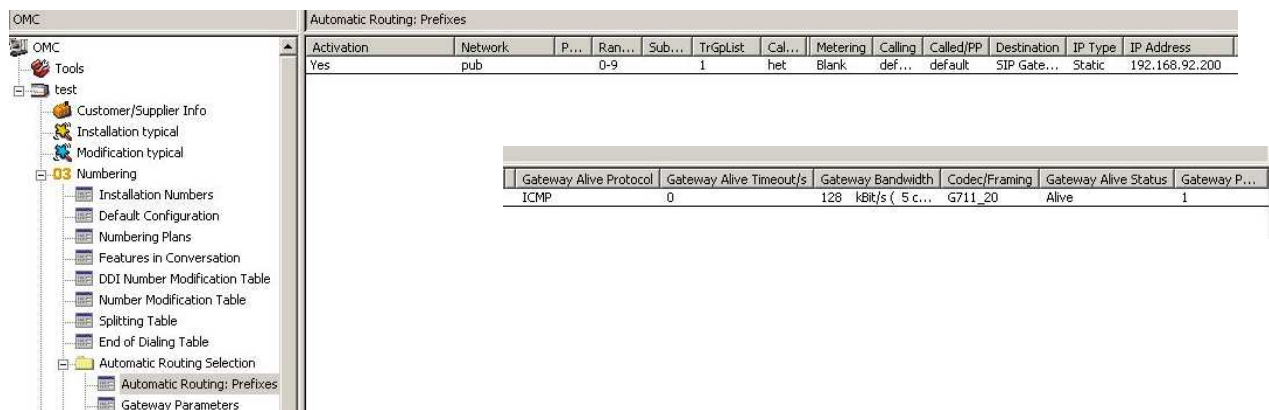


Check Public numbering plan for incoming calls.

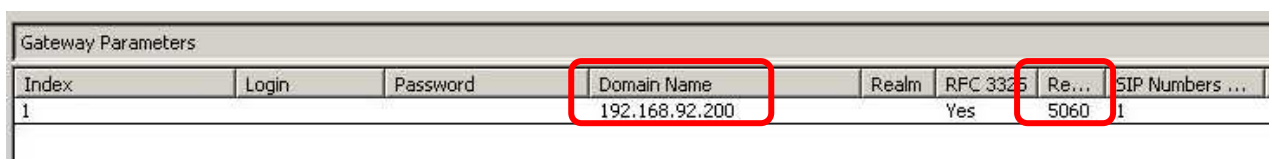
At Trunk Groups List assign List ID with Index at menu **Numbering** → **Automatic Routing Selection** → **Trunk Groups List**



Set up IP address of 2N® StarGate, codecs, bandwidth and GW keep alive timeout for VoIP trunk at menu **Numbering** → **Automatic Routing Selection** → **Automatic Routing: Prefixes**



At menu **Numbering** → **Automatic Routing Selection** → **Gateway Parameters** set up listening port 5060. If you want to use 2N® StarGate with different listening port (for example 5065), just setup the number of the port here.





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