



2N<sup>®</sup>

# VoiceBlue Next



## **2N<sup>®</sup> VoiceBlue Next & Cisco Unified Communications Manager Express verze 4.1(0)**

connected via SIP trunk

Quick guide

Version 2.00

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

## 2N® VoiceBlue Next has these parameters:

- IP address 192.168.22.42
- Incoming port: 5060

## Cisco Unified Communication Manager Express parameters:

- IP address 192.168.22.35
- Incoming port: 5060

## SIP TRUNK INTERCONNECTION

- 1) For the setting of the trunk between the VoiceBlue Next and your PBX you need to configure SIP proxy (GSM→IP) for GSM incoming calls. SIP proxy (IP→GSM) is designed for secure communication just with traffic from your CME. 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 displays the configuration page for a 2N Gateway. The interface includes a sidebar with navigation options like 'Gateway control', 'Gateway configuration', and 'Configuration backup'. The main content area is titled 'Gateway' and contains several configuration sections. Two callout boxes provide context: one points to the 'SIP proxy (GSM->IP)' field, and another points to the 'SIP proxy (IP->GSM)' field.

Section	Parameter	Value	Port	Action
Codec settings	G711:	2 x 10ms		<input type="checkbox"/>
	G729:	2 x 10ms		<input type="checkbox"/>
	Codec priority	Priority 1:	G711a (8)	
IP addresses	SIP proxy (IP->GSM):	0.0.0.0	5060	Set default port
	SIP proxy (GSM->IP):	192.168.22.35	5060	Set default port
	SIP registrar:	0.0.0.0	5060	Set default port
	NAT firewall:	0.0.0.0		
	STUN server:	0.0.0.0	3478	Set default port

## 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.

The screenshot displays the configuration interface for a 2N Gateway. The interface is divided into several sections:

- Header:** Features the 2N TELECOMMUNICATIONS logo on the left and the Gateway logo with a mobile phone icon on the right. Below the Gateway logo are the options "Gateway | Update | Restart".
- Left Sidebar:** Contains a "Gateway control" section and a "Gateway configuration" section with a list of menu items: System parameters, VoIP parameters, GSM basic parameters, GSM groups assignment, GSM outgoing groups, GSM incoming groups, **Prefixes** (highlighted), LCR table, CLIP Routing table, Mobility Extension, Ethernet configuration, Login configuration, Web configuration, and Report configuration. Below this is a "Configuration backup" section.
- Main Content Area:**
  - Prefixes:** A title for the main configuration area.
  - GSM prefix lists:** A row of tabs labeled "Prefixlist 1" through "Prefixlist 8".
  - Basic settings:** Includes "GSM network ID:" with an empty text input field and "Default count of digits:" with a dropdown menu set to "9".
  - Table of replaced prefixes:** A section with the note "Only 0123456789\*#+ characters are allowed". It contains an empty list box, a "Prefix:" input field, a "Replace with:" input field, and "Add", "Remove", and "Remove all" buttons.
  - Table of accepted prefixes:** A section with the note "Only 0123456789\*#+ characters are allowed". It contains a list box with "6" and "7" selected, a "Prefix:" input field, a "[Digits count]:" dropdown menu, and "Add", "Remove", and "Remove all" buttons.
- Footer:** Includes a "Logout" button with a help icon and three document icons (one with a checkmark, one with an X, and one with a checkmark).

- 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 cards below to which GSM outgoing group.

The screenshot shows the 2N Gateway web interface. The top left features the 2N TELECOMMUNICATIONS logo. The top right has the Gateway logo and navigation links: Gateway | Update | Restart. A left sidebar contains a 'Gateway control' section with a 'Gateway configuration' menu listing various settings like System parameters, VoIP parameters, GSM basic parameters, GSM groups assignment, GSM outgoing groups, GSM incoming groups, Prefixes, LCR table, CLIP Routing table, Mobility Extension, Ethernet configuration, Login configuration, Web configuration, and Report configuration. Below this is a 'Configuration backup' section and a 'Logout' button. The main content area is titled 'LCR table' and contains a table with the following data:

Prefix list	Time limitation	Weekend usage	Max. length of call	Groups	Add	Remove all
1/	0:00/24:00	Use as in week	Off	1	Edit	Remove
2/	0:00/24:00	Use as in week	Off	2	Edit	Remove

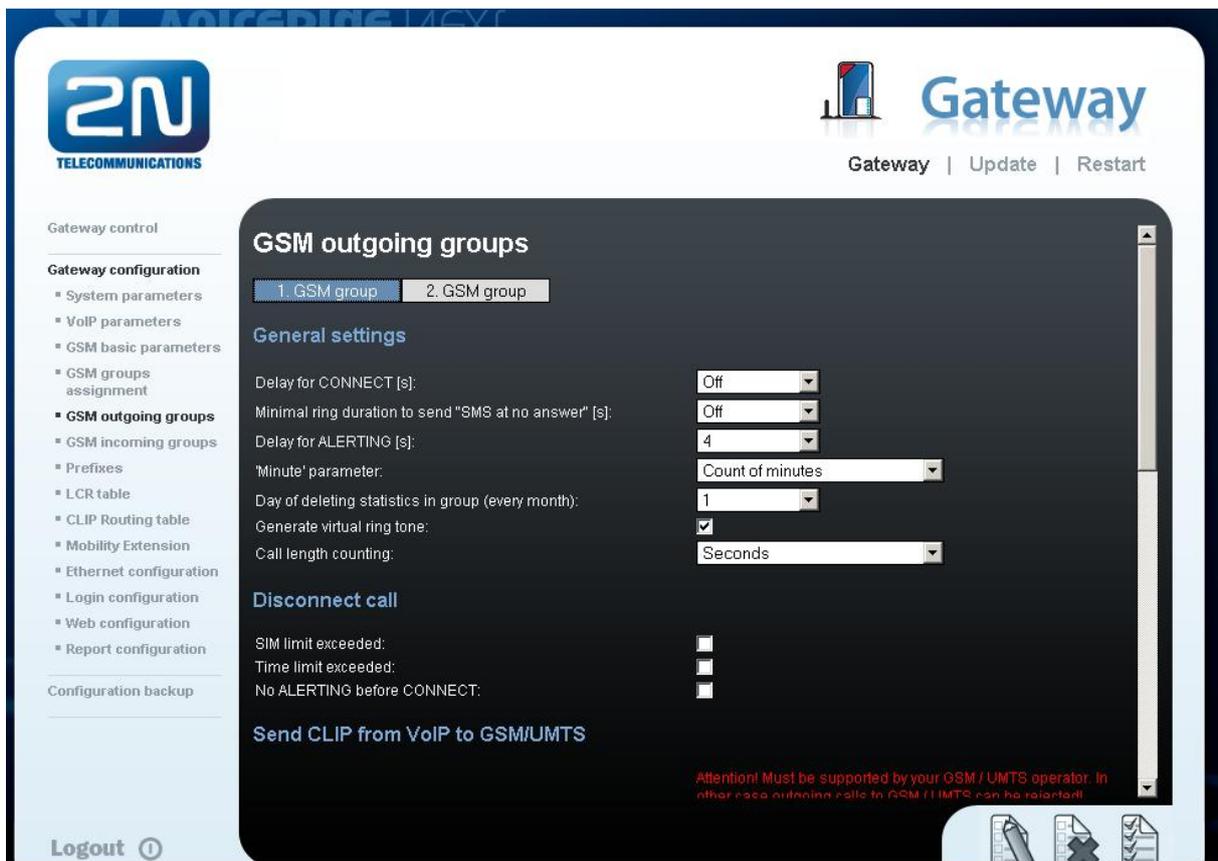
At the bottom right of the table area, there are three icons: a pencil (edit), a document with a cross (delete), and a document with a checkmark (save).

The screenshot shows the 'GSM groups assignment' configuration page. It features a left sidebar with 'Gateway control' and 'Gateway configuration' (System parameters, VoIP parameters, GSM basic parameters). The main content area is titled 'GSM groups assignment' and contains the following configuration fields:

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

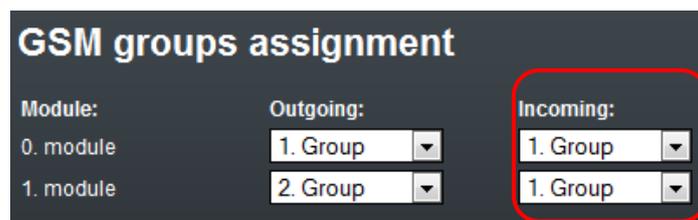
#### 4) Configuration of GSM outgoing groups:

You are able to set up different setting for each GSM group (CLIR, free minutes, Virtual ring tone, roaming and others)



#### 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.



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.

The screenshot shows the 'Gateway' configuration interface. At the top left is the '2N TELECOMMUNICATIONS' logo. At the top right is the 'Gateway' logo with a mobile phone icon and the text 'Gateway | Update | Restart'. On the left side, there is a navigation menu with categories: 'Gateway control', 'Gateway configuration' (with sub-items like System parameters, VoIP parameters, GSM basic parameters, GSM groups assignment, GSM outgoing groups, GSM incoming groups, Prefixes, LCR table, CLIP Routing table, Mobility Extension, Ethernet configuration, Login configuration, Web configuration, Report configuration), and 'Configuration backup'. The main content area is titled 'GSM incoming groups' and has two tabs: '1. GSM group' (selected) and '2. GSM group'. Under 'General settings', there are several configuration fields: 'Mode' (dropdown menu set to 'Accept incoming calls + dialtone'), 'Minimum digits in DTMF' (dropdown menu set to '4'), 'Maximum digits in DTMF' (dropdown menu set to '9'), 'DTMF dialling timeout [s]' (dropdown menu set to '10'), 'Day of deleting GSM inc. group statistics (every month):' (dropdown menu set to '1'), 'Prefix before DISA dial-in:' (text input field), 'CLIP (\* removes one digit):' (text input field), and 'Looping of voice message [min]:' (dropdown menu set to 'Off'). Below this is the 'Send CLIP from GSM/UMTS to VoIP' section with fields for 'Transfer CLIP from GSM/UMTS:' (checkbox), 'Separating char:' (text input field), and 'Modify (\* removes one digit):' (text input field). At the bottom right of the main area are icons for edit, delete, and refresh. At the bottom left is a 'Logout' button.

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 100 to your CME (if you set up SIP proxy (GSM->IP) in VoIP parameters).

The screenshot shows the 'List of called numbers' configuration interface. At the top left is the title 'List of called numbers'. Below it is a text box containing the number '100'. Above the text box is the instruction 'Only 0123456789\*#+ characters are allowed'. To the right of the text box are three buttons: 'Add', 'Remove', and 'Remove all'. At the bottom right of the main area are icons for edit and delete. At the bottom left is a 'Logout' button.

# CISCO UNIFIED COMMUNICATION MANAGER EXPRESS SETTING

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For configuration was used freeware program - Cisco Configuration Professional - version 2.3.

For configuration, enter the menu: **Configure** → **Voice** → **Dial Plan** → **VoIP** and set up the prefix and IP address to the 2N VoiceBlue Next gateway.

The prefix **6.T** as in the example means that prefix is 6 plus other digits after 6 without limit.

The screenshot shows the 'Calling Restrictions' tab of a dial peer configuration window. The fields are as follows:

- Dial Peer Number \*:** 2
- Description :** 2N\_VoiceBlue\_Next (1-64 Characters)
- Priority :** Priority 0
- Remote Site :**  192.168.22.42  SIP Trunk
- Destination Number :** 6.T
- Incoming Called Number :** (empty)
- Answer Address :** (empty)
- Shutdown Dial Peer :**  No  Yes
- Protocol :**  H.323  SIP
- Codec :**  g711alaw  Voice Class Codec 1
- DTMF Tone Relay Type :** rtp-nte

**Voice Activity Detection**

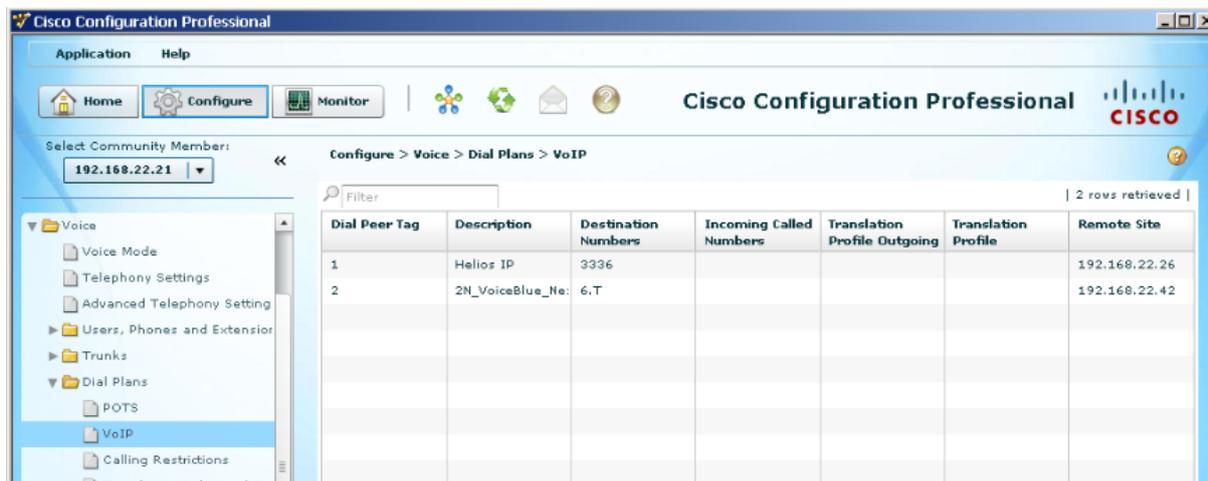
With voice activity detection enabled, only speech voice data packets are sent over the network, and silence voice data packets are dropped optimizing the network bandwidth usage.

Enable voice activity detection

\* Indicates a mandatory field

OK Cancel

In the picture below, you can see the configuration program with the saved routing to the 2N VoiceBlue Next.



Incoming calls are automatically enabled by a new trunk. All incoming calls to from 2N VoiceBlue Next will be routed to stations in CME or you can create your own dial plan.

In the CME version 4.1 you are not able to register SIP phones.



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