



2N[®]

VoiceBlue Next



2N[®] VoiceBlue Next & 3CX PBX

connected via SIP trunk

Quick guide

Version 1.00

www.2n.cz

2N® VoiceBlue Next has these parameters:

- IP address 192.168.50.51
- Incoming port: 5060
- Firmware version: 01.00.03rc3

3CX PBX parameters:

- IP address 192.168.50.115
- Incoming port: 5060
- Software version: 9.0

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 CCM. 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 gateway G729. On the left is a navigation menu with categories like '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', 'Ethernet configuration', 'Login configuration', 'Web configuration', and 'Report configuration'. The main content area is divided into several sections:

- Codec priority:** Lists three priorities with dropdown menus: Priority 1 (G711a (8)), Priority 2 (G711u (0)), and Priority 3 (G729 (18)).
- IP addresses:** A table with four rows for SIP proxy (IP->GSM), SIP proxy (GSM->IP), SIP registrar, and NAT firewall. Each row has an IP address field, a port field, and a 'Set default port' button. The SIP proxy (GSM->IP) row is highlighted.
- Tones generated to VoIP:** A dropdown menu for 'Ring tone to VoIP' set to 'From GSM'.

Two callout boxes are present:

- A callout box pointing to the IP address field '0.0.0.0' in the SIP proxy (GSM->IP) row, containing the text: 'The IP address to which the traffic is send from GSM'.
- A callout box pointing to the port field '5060' in the same row, containing the text: 'The IP address and port which will accept traffic from'.

At the bottom left is a 'Logout' button with an information icon. At the bottom right are three icons: a pencil, a document with a red X, and a document with a checkmark.

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 'Gateway' configuration interface for LCR. The interface includes a sidebar with navigation options and a main content area for configuring GSM prefix lists.

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
- Ethernet configuration
- Login configuration
- Web configuration
- Report configuration

Configuration backup

Logout ⓘ

Gateway | Update | Restart

Prefixes

GSM prefix lists

Prefixlist 1 | Prefixlist 2 | Prefixlist 3 | Prefixlist 4 | Prefixlist 5 | Prefixlist 6 | Prefixlist 7 | Prefixlist 8

Basic settings

GSM network ID:

Default count of digits:

Table of replaced prefixes

Only 0123456789*#+ characters are allowed

| Prefix |
|--------|
| / |

Prefix:

Replace with:

Add

Remove

Remove all

Table of accepted prefixes

Only 0123456789*#+ characters are allowed

| Prefix |
|--------|
| 6 |
| 7 |

Prefix:

[Digits count]:

Add

Remove

Remove all

- 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 'Gateway control' 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 under 'Gateway control' with 'Gateway configuration' expanded to show various options, including 'LCR table' which is currently selected. The main content area displays the 'LCR 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 |

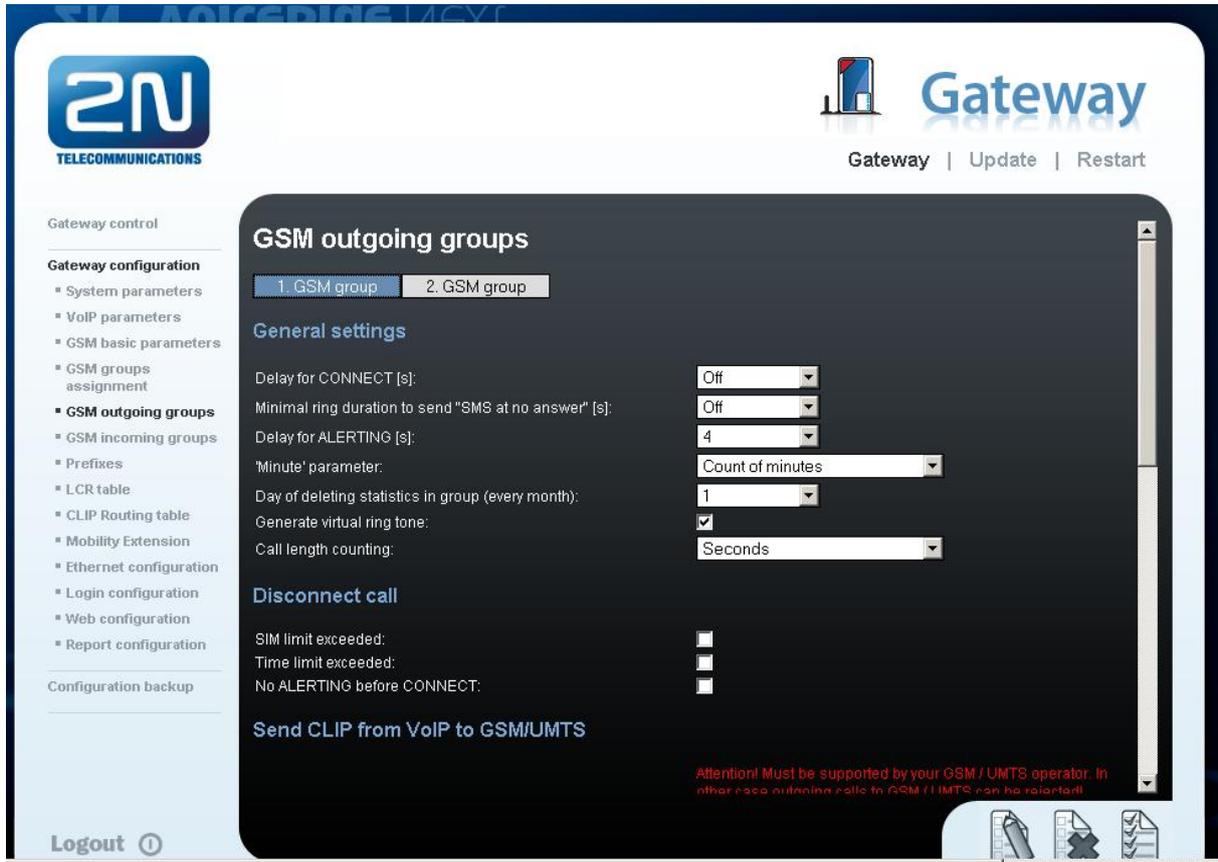
At the bottom left, there is a 'Logout' button with an information icon. At the bottom right, there are three icons representing a clipboard, a document with a cross, and a document with a checkmark.

The screenshot shows the 'GSM groups assignment' configuration screen. On the left, the 'Gateway control' menu is visible with 'Gateway configuration' expanded to 'GSM basic parameters'. The main content area is titled 'GSM groups assignment' and contains the following configuration options:

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

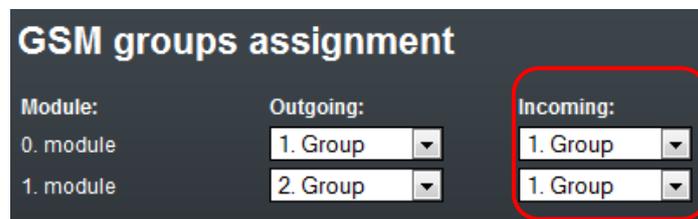
4) Configuration of GSM outgoing groups:

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.

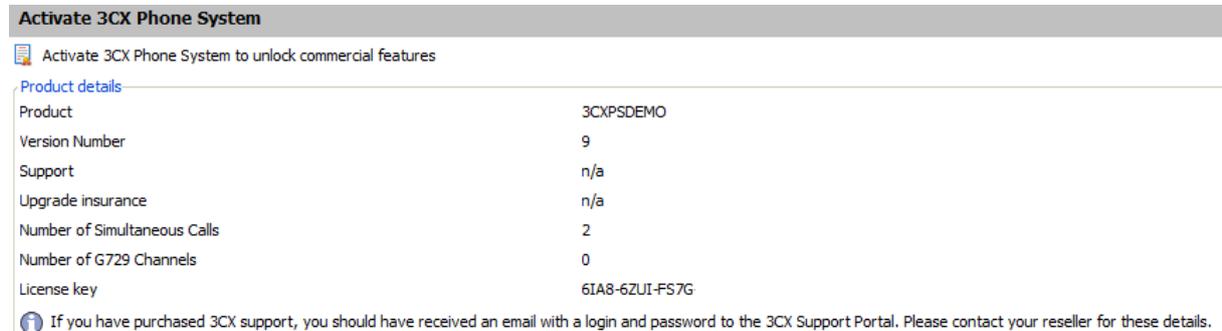
The screenshot shows the 'Gateway' configuration interface. At the top left is the '2N TELECOMMUNICATIONS' logo. At the top right is the 'Gateway' logo with 'Update' and 'Restart' links. The main content area is titled 'GSM incoming groups' and has two tabs: '1. GSM group' (selected) and '2. GSM group'. Under 'General settings', the 'Mode' is set to 'Accept incoming calls + dialtone'. Below this, several parameters are configured: 'Minimum digits in DTMF' is 4, 'Maximum digits in DTMF' is 9, 'DTMF dialling timeout [s]' is 10, and 'Day of deleting GSM inc. group statistics (every month)' is 1. There are also input fields for 'Prefix before DISA dial-in', 'CLIP (-' removes one digit)', and 'Looping of voice message [min]' set to 'Off'. A section titled 'Send CLIP from GSM/UMTS to VoIP' includes a checkbox for 'Transfer CLIP from GSM/UMTS', a 'Separating char.' field, and a 'Modify (-' removes one digit)' field. The left sidebar contains a navigation menu with categories like 'Gateway control', 'Gateway configuration', and 'Configuration backup'. At the bottom left is a 'Logout' button, and at the bottom right are icons for editing and saving.

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 3CX PBX (if you set up SIP proxy (GSM->IP) in VoIP parameters).

The screenshot shows the 'List of called numbers' configuration interface. At the top, it states 'Only 0123456789*#+ characters are allowed'. Below this is a list box containing the number '100'. To the right of the list box are three buttons: 'Add', 'Remove', and 'Remove all'. At the bottom right, there are icons for editing and saving.

3CX PBX SETTING

You can download the 3CX PBX for free. Then you can activate a free version by registering the product and using a demo license key (**Settings → Activate License**) and the 3CX system will be changed to Commercial Edition (3CXPSDEMO) where all features are available. Demo license is not limited by time.



| Product details | |
|------------------------------|----------------|
| Product | 3CXPSDEMO |
| Version Number | 9 |
| Support | n/a |
| Upgrade insurance | n/a |
| Number of Simultaneous Calls | 2 |
| Number of G729 Channels | 0 |
| License key | 6IA8-6ZUI-FS7G |

ⓘ If you have purchased 3CX support, you should have received an email with a login and password to the 3CX Support Portal. Please contact your reseller for these details.

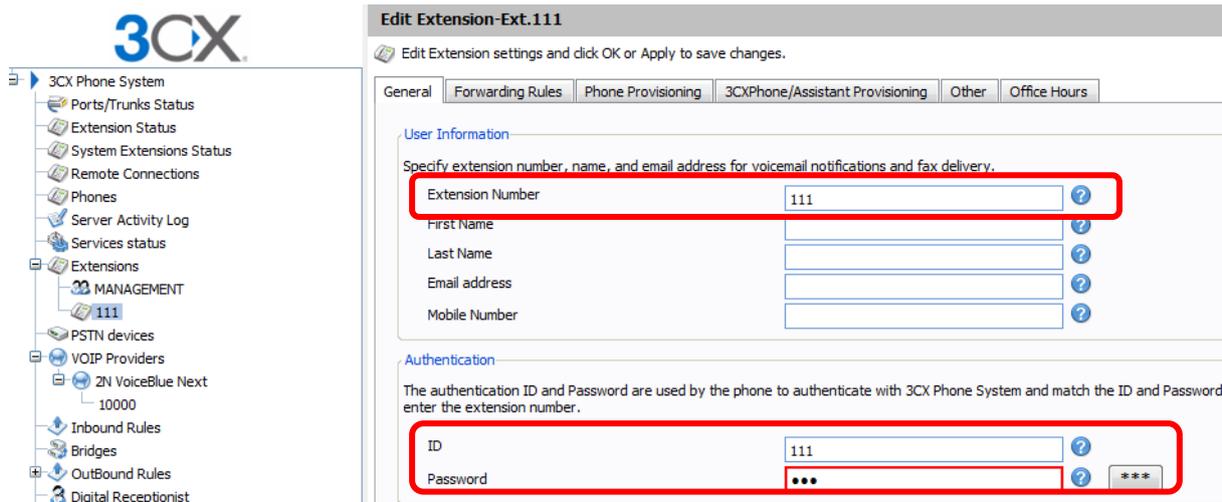
Differences between Free version and Commercial Edition you can check in this document on the page 11:

http://www.3cx.com/phone-system/3CXPhoneSystem_brochure.pdf

In the Commercial Edition you can use only 2 concurrent SIP calls, use codec G.729, use a SIP trunk, etc...

1) Create an Extension

Use the button Add Extension and fill up Extension Number and ID and Password.



2) Create a new SIP trunk

Add → VOIP Provider in the menu of the 3CX PBX. Use the name of provider and choose **Generic SIP Trunk**. Then use the button **NEXT**.

VOIP Providers

 Add VOIP Provider Wizard

Add VOIP Provider Wizard

Name of Provider 

Choose a Provider:

-  [Actio.pl](#) PL
-  [Broadvox Fusion \(IP Based\)](#) US
-  [Broadvox Fusion \(Register\)](#) US
-  [CallCentric](#) US
-  [Cbeyond](#) Worldwide
-  [CellIP](#) SE
-  [EasyCall](#) GR
-  [Engin](#) AU
-  [G7Eleven](#) IE
-  [Generic SIP Trunk](#)
-  [Generic VoIP Provider](#)

Fill up the IP address and the listening port of the 2N® VoiceBlue Next.

VOIP Providers

 Add VOIP Provider Wizard

VOIP Provider Details:

Enter the hostname and port for your VOIP Provider's SIP Server

| | | |
|---------------------------------------|--|---|
| SIP server hostname or IP | <input type="text" value="192.168.50.51"/> |  |
| SIP Server port | <input type="text" value="5060"/> |  |
| Outbound proxy hostname or IP | <input type="text"/> |  |
| Outbound proxy port (default is 5060) | <input type="text" value="5060"/> |  |

Fill up the External number (this number will be identification of call – FROM and CONTACT field)

VOIP Providers

Add VOIP Provider Wizard

Account Details

Enter the Authentication ID, Password and number of your account

| | | |
|-------------------------|-------|---|
| External Number | 10000 | ? |
| Authentication ID | 10000 | ? |
| Authentication Password | •••• | ? |

Simultaneous Calls

| | | |
|----------------------------|---|---|
| Maximum simultaneous calls | 2 | ? |
|----------------------------|---|---|

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Set up the the extension where incoming call will be routed from the 2N® VoiceBlue Next.

VOIP Providers

Add VOIP Provider Wizard

Office Hours

Configure where calls should be routed during office hours.

End Call

Connect to Extension 111 ?

Connect to Queue / Ring Group ?

Connect to Digital Receptionist ?

Voicemail box for Extension 111 ?

Forward to Outside Number ?

Send fax to email of extension email of extension 888 ?

Same as Out of Office hours

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Set up Outgoing rules :

- 1) Prefix called from the extension
- 2) Set up the range of extension which can use this trunk (e.g. 111-120)
- 3) SIP trunk and Strip Digits (0 will be striped in this case)

Edit Outbound Rule

Create an Outbound Call Rule to configure on which PSTN port, VOIP provider or bridge an outbound calls should be placed on

General

Rule Name Rule for 2N VoiceBlue Next ?

Apply this rule to these calls

Define to which outbound calls the rule must apply

| | | |
|---|-----|---|
| Calls to numbers starting with (Prefix) | 07 | ? |
| Calls from extension(s) | 111 | ? |
| Calls to Numbers with a length of | | ? |

Make outbound calls on

Configure up to 3 routes for calls. The second and third route will be used as backup. For each route, digits can be stripped or added.

| Route | | Strip Digits | Prepend |
|-------|-------------------|--------------|---------|
| 1 | 2N VoiceBlue Next | 1 | |
| 2 | | 1 | |
| 3 | | 1 | |

OK Cancel Apply

Set up codecs and turn off registration of the SIP trunk. This setting is in **VOIP Providers → Advanced**

Edit VOIP Provider settings and click OK or Apply to save changes

General | **Advanced** | Outbound Parameters | Inbound Parameters | Source ID | DID

Provider Capabilities
Configure options related to the SIP capabilities of your provider

- Supports Re-Invite ?
- Supports 'Replace' ?
- PBX Delivers Audio ?
- Switch on Secure RTP (SRTP) ?

Registration Settings
Configure options related to the SIP capabilities of your provider

Time between registration attempts (in seconds) ?

Require registration for: ?

Which IP to use in 'Contact' field for registration:

- External(STUN resolved) ?
- Internal ?
- Specified IP

Codec priorities
Specify which codecs to use and according to which priority.

Available Codecs

| |
|-------|
| Speex |
| iLBC |
| G729 |

Assigned Codecs

| |
|-------------|
| G.711 U-law |
| G.711 A-law |
| GSM-FR |

3) Create inbound rules

- specify numbers or range numbers what could be dialed from 2N® VoiceBlue Next. You can use "*" for all numbers.
- specify the SIP trunk from which the number will come

- specify the extension, ring group or voicemail where the call will be connected

4) Make a call

You can register your SIP phone or download the 3CXPhone from 3CX webpage for free:

<http://www.3cx.com/VOIP/voip-phone.html>

Register your SIP phone to the 3CX PBX and make an outgoing call with specified prefix to GSM via

2N® VoiceBlue Next.



2N TELEKOMUNIKACE a.s.

Modřanská 621, 143 01 Praha 4
 tel.: 261 301 111, fax: 261 301 999,
 e-mail: sales@2n.cz
 www.2n.cz