



2N<sup>®</sup>

# VoiceBlue Next



## **2N<sup>®</sup> VoiceBlue Next & 3CX PBX**

connected via SIP trunk

Quick guide

Version 1.00

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

**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

**G729:** 2 x 10ms

**Codec priority**

Priority 1: G711a (8)  
Priority 2: G711u (0)  
Priority 3: G729 (18)

**IP addresses**

SIP proxy (IP->GSM):	0.0.0.0	: 5060	Set default port
SIP proxy (GSM->IP):	192.168.50.115	: 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
Next STUN server request (60-6553, 0=off) [s]:	600		

**Tones generated to VoIP**

Ring tone to VoIP: From GSM

Logout

## 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 2N Gateway configuration web interface. The top left features the 2N TELECOMMUNICATIONS logo. The top right shows the Gateway logo with 'Update' and 'Restart' links. A left sidebar contains a 'Gateway control' section and a 'Gateway configuration' menu with options 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. The main content area is titled 'Prefixes' and includes a 'GSM prefix lists' tab bar with eight tabs (Prefixlist 1 to 8). Below this is the 'Basic settings' section with fields for 'GSM network ID' and 'Default count of digits' (set to 9). Two tables are present: 'Table of replaced prefixes' and 'Table of accepted prefixes', both with a warning 'Only 0123456789\*#+ characters are allowed'. Each table has a list of prefixes and associated actions (Prefix, Replace with, Add, Remove, Remove all). The 'Table of accepted prefixes' shows prefixes 6 and 7. At the bottom left is a 'Logout' button, and at the bottom right are three icons representing different configuration files.

**2N TELECOMMUNICATIONS**

**Gateway**  
Gateway | Update | Restart

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

**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	Replace with
--------	--------------

Add Remove Remove all

**Table of accepted prefixes**


Only 0123456789\*#+ characters are allowed


Prefix	[Digits count]
6	
7	

Add Remove Remove all

Logout ⓘ

- 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.





# Gateway

Gateway | Update | Restart

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

## LCR table

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



Logout ⓘ

Gateway control

Gateway configuration

- System parameters
- VoIP parameters
- GSM basic parameters

## GSM groups assignment

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)

The screenshot shows the 'Gateway control' interface for '2N TELECOMMUNICATIONS'. The main section is titled 'GSM outgoing groups' and has two tabs: '1. GSM group' and '2. GSM group'. Under 'General settings', there are several configuration options: 'Delay for CONNECT [s]' (Off), 'Minimal ring duration to send "SMS at no answer" [s]' (Off), 'Delay for ALERTING [s]' (4), 'Minute' parameter (Count of minutes), 'Day of deleting statistics in group (every month)' (1), 'Generate virtual ring tone' (checked), and 'Call length counting' (Seconds). Under 'Disconnect call', there are three checkboxes: 'SIM limit exceeded', 'Time limit exceeded', and 'No ALERTING before CONNECT'. Below this is a section 'Send CLIP from VoIP to GSM/UMTS'. At the bottom right, there is a red warning message: 'Attention! Must be supported by your GSM / UMTS operator. In other case outgoing calls to GSM / UMTS can be rejected!'. The left sidebar contains a 'Gateway configuration' menu with options 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'. There is also a 'Configuration backup' link and a 'Logout' button.

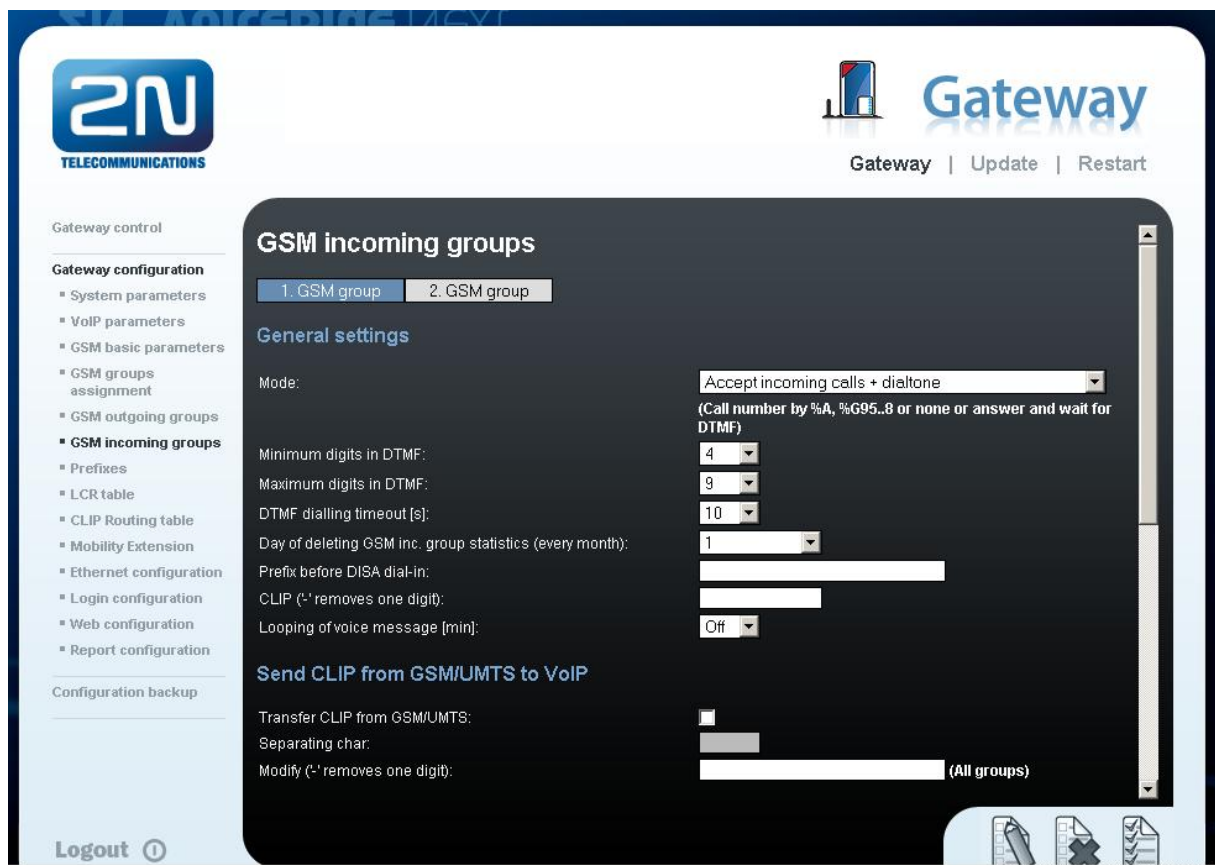
#### 5) Incoming calls

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.

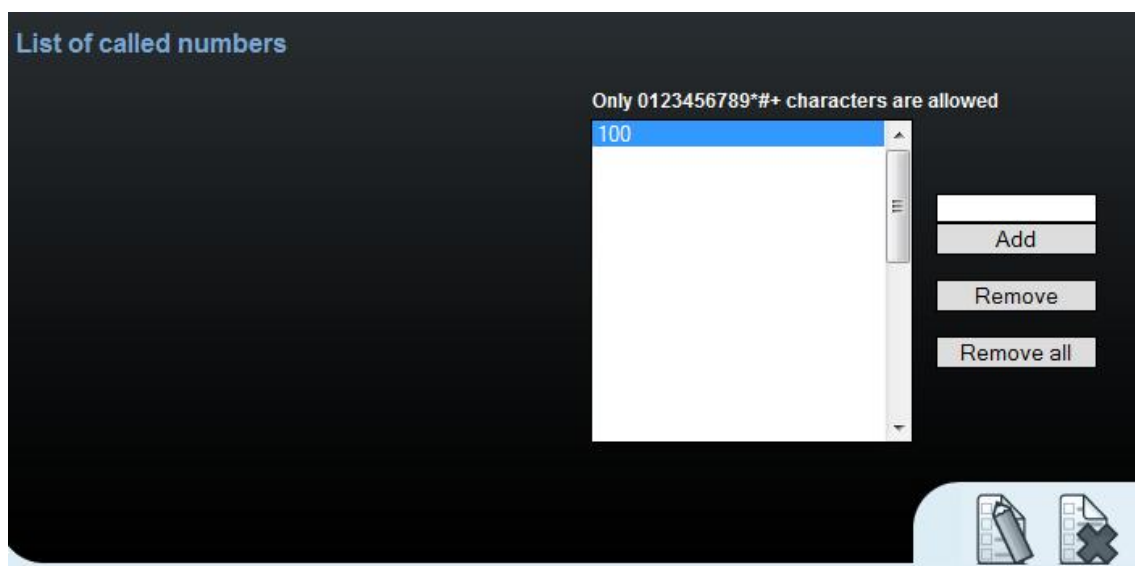
The screenshot shows the 'GSM groups assignment' interface. It has a table with three columns: 'Module:', 'Outgoing:', and 'Incoming:'. The 'Outgoing:' column has two dropdown menus: '1. Group' and '2. Group'. The 'Incoming:' column has two dropdown menus: '1. Group' and '1. Group'. The 'Incoming:' column is highlighted with a red box.

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

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.



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).



# 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.

**Activate 3CX Phone System**

Activate 3CX Phone System to unlock commercial features

**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

**i** 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](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.

**3CX**

3CX Phone System

- Ports/Trunks Status
- Extension Status
- System Extensions Status
- Remote Connections
- Phones
- Server Activity Log
- Services status
- Extensions
  - MANAGEMENT
  - 111
- PSTN devices
- VOIP Providers
  - 2N VoiceBlue Next
    - 10000
- Inbound Rules
- Bridges
- OutBound Rules
- Digital Receptionist

**Edit Extension-Ext.111**

Edit Extension settings and click OK or Apply to save changes.

General Forwarding Rules Phone Provisioning 3CXPhone/Assistant Provisioning Other Office Hours

**User Information**

Specify extension number, name, and email address for voicemail notifications and fax delivery.

Extension Number 111 ?

First Name ?

Last Name ?

Email address ?

Mobile Number ?

**Authentication**

The authentication ID and Password are used by the phone to authenticate with 3CX Phone System and match the ID and Password enter the extension number.

ID 111 ?

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:

<input type="radio"/>		<a href="#">Actio.pl</a>	PL
<input type="radio"/>		<a href="#">Broadvox Fusion (IP Based)</a>	US
<input type="radio"/>		<a href="#">Broadvox Fusion (Register)</a>	US
<input type="radio"/>		<a href="#">CallCentric</a>	US
<input type="radio"/>		<a href="#">Cbeyond</a>	Worldwide
<input type="radio"/>		<a href="#">CellIP</a>	SE
<input type="radio"/>		<a href="#">EasyCall</a>	GR
<input type="radio"/>		<a href="#">Engin</a>	AU
<input type="radio"/>		<a href="#">G7Eleven</a>	IE
<input checked="" type="radio"/>		<a href="#">Generic SIP Trunk</a>	
<input type="radio"/>		<a href="#">Generic VoIP Provider</a>	

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"/>	

< Back

Next >

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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< Back    Next >

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

< Back    Next >

### 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

Add >

< Remove

Assigned Codecs

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

Up

Down

### 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

**3CX**

3CX Phone System

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- OutBound Rules
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- Ring Groups
- Call Queues
- Fax Machines
- Settings
  - 3CX Phone System Updates
- Links
- Help

**Add DID**

Route calls to DID/DDI numbers directly to an extension

**DID/DDI Name**

Enter a DID or string to look for in the SIP "to" field. Use wildcards (\*) to match any digit for that entry. For example, entries 22444032 OR 2244403\* will both match calls with a dialed number of +35722444032 in the "to" field

DID/DDI Name: All calls

**DID/DDI number/mask**

Enter a Mask for this DID. You can use the \* character either before or after your mask.

DID/DDI number/mask: \*

**Apply this rule to these ports**

Select the Gateway you want this DID/DDI rule to be applied to. You can select on the whole gateway which will apply the rule to all the ports, or you can select individual ports.

Available ports: 2N VoiceBlue Next

**Office Hours**

Configure where calls to this DID/DDI should be routed during office hours.

End Call

Connect to Extension: 111

## 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.



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