

3CX - How to interconnect with 3CX?

- 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

2N[®] Voice Blue Next settings

1. SIP trunk interconnection

For the setting of the trunk between the 2N[®] VoiceBlue Next and your PBX you need to configure SIP proxy (GSM→IP) for GSM incoming calls. SIP proxy (IP→GSM) is designed to secure communication with traffic from your CCM only. You can specify the IP address and port from which SIP packets will be accepted.

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



The screenshot shows the 'Gateway configuration' page for a 2N Gateway. The 'VoIP parameters' section is expanded to show 'SIP proxy' settings. Two callouts are present:

- Callout 1:** Points to the 'SIP proxy (IP->GSM)' field, which is set to '192.168.92.246'. The text says: "The IP address to which the traffic is send".
- Callout 2:** Points to the 'SIP proxy (GSM->IP)' field, which is set to '192.168.92.246' and port '5060'. The text says: "The IP address and port from which SIP packets will be accepted".

Other visible settings include:

- SIP proxy (GSM->IP): 192.168.92.246 : 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: English

2. Configuration of the LCR (Least Cost Routing)

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

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

Logout ⓘ

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: 9

Table of replaced prefixes

Only 0123456789*#+ characters are allowed

Prefix	Replace with
/	

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

You need to create LCR rule for defined prefixes. The GSM group defines a way for the outgoing call routing. An appropriate SIM card is selected based on the GSM groups assignment.



Gateway control

Gateway configuration

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- GSM incoming groups
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- LCR table**
- CLIP Routing table
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- Report configuration

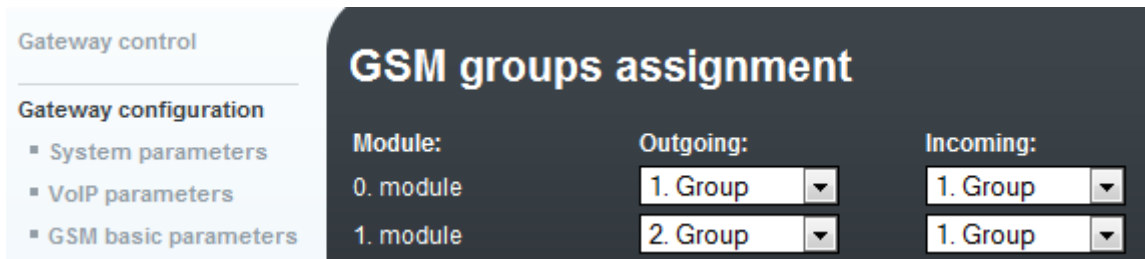
Configuration backup

Logout ⓘ

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

Gateway | Update | Restart



Gateway control

Gateway configuration

- System parameters
- VoIP parameters
- GSM basic parameters

GSM groups assignment

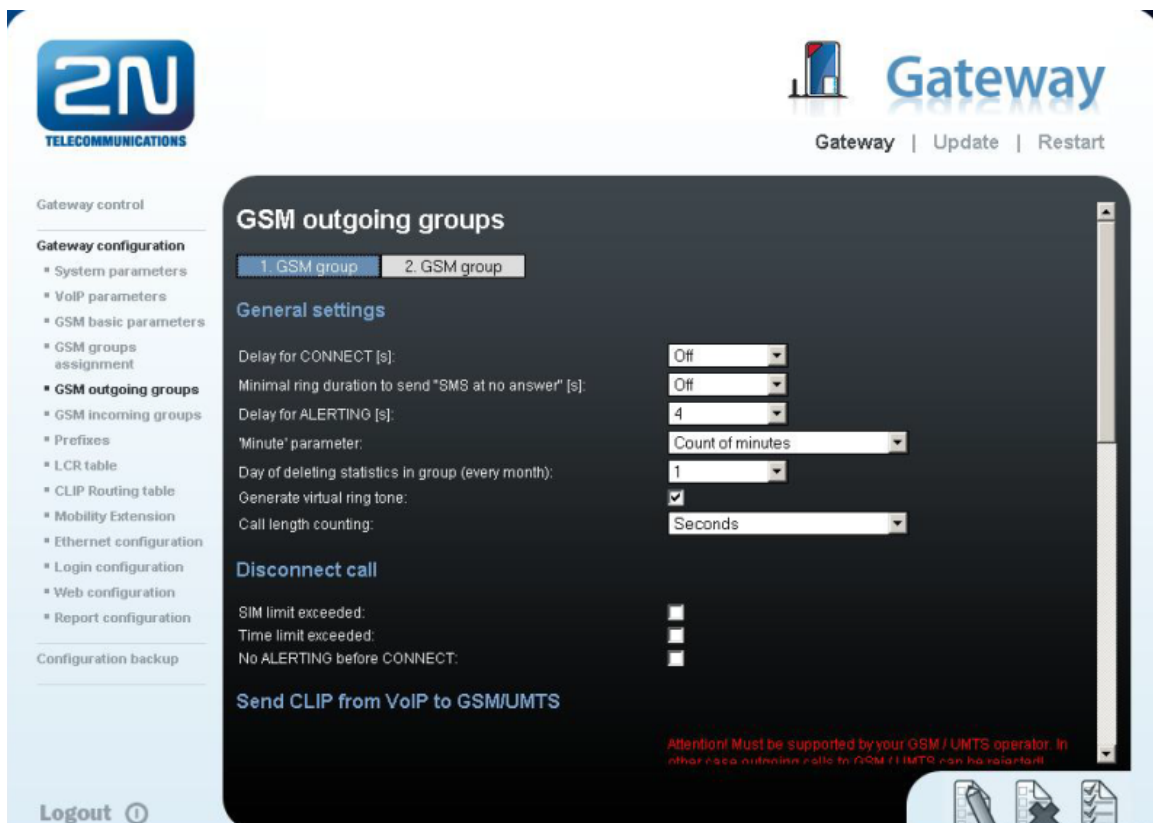
Module: Outgoing: Incoming:

0. module 1. Group 1. Group

1. module 2. Group 1. Group

3. Configuration of GSM outgoing groups

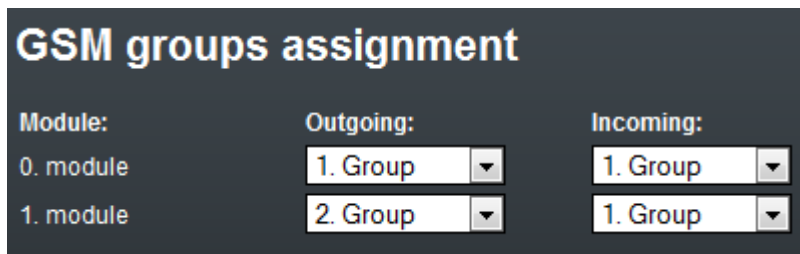
You are able to set up different settings for each GSM group (CLIR, free minutes, virtual ring tone, roaming and others)



The screenshot shows the 'Gateway' configuration page for 'GSM outgoing groups'. The interface includes a sidebar with navigation options like 'Gateway control', 'Gateway configuration', and 'Configuration backup'. The main content area is titled 'GSM outgoing groups' and has two tabs: '1. GSM group' (selected) and '2. GSM group'. Under 'General settings', there are several dropdown menus and checkboxes for parameters such as 'Delay for CONNECT [s]', 'Minimal ring duration to send "SMS at no answer" [s]', 'Delay for ALERTING [s]', 'Minute' parameter, 'Day of deleting statistics in group (every month)', 'Generate virtual ring tone', and 'Call length counting'. A 'Disconnect call' section contains three checkboxes for 'SIM limit exceeded', 'Time limit exceeded', and 'No ALERTING before CONNECT'. At the bottom, there is a checkbox for 'Send CLIP from VoIP to GSM/UMTS' and a red warning message: 'Attention! Must be supported by your GSM / UMTS operator. In other case outgoing calls to GSM / UMTS can be rejected!'. A 'Logout' button is visible in the bottom left corner.

4. 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' configuration table. It has three columns: 'Module:', 'Outgoing:', and 'Incoming:'. The rows represent different modules and their assigned groups.

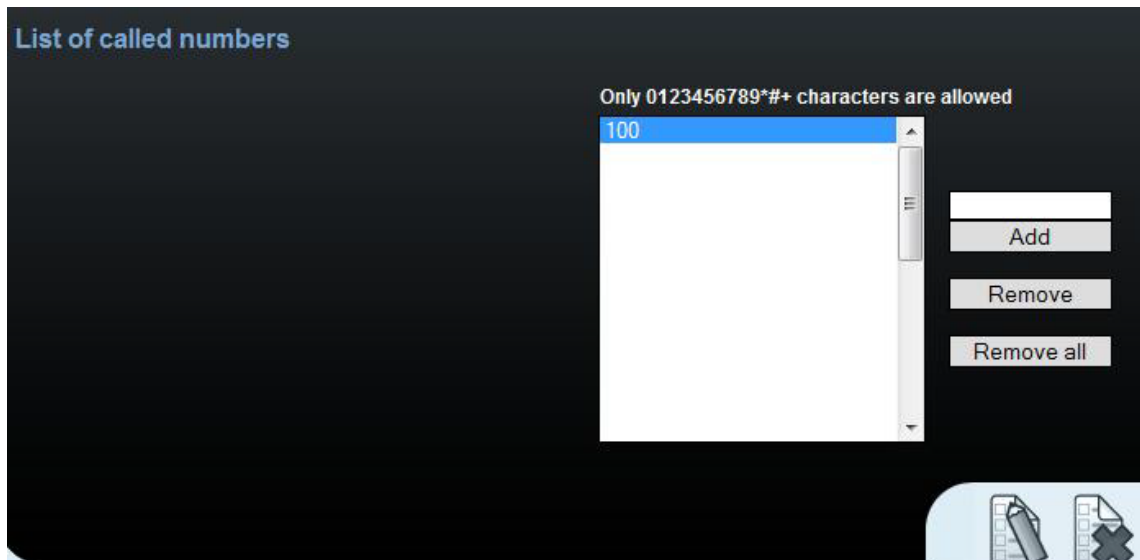
Module:	Outgoing:	Incoming:
0. module	1. Group	1. Group
1. module	2. Group	1. Group

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.



The screenshot shows the 'Gateway' configuration page for 'GSM incoming groups'. The interface includes a sidebar with navigation options like 'Gateway control', 'Gateway 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', the 'Mode' is set to 'Accept incoming calls + dialtone'. Other settings include: Minimum digits in DTMF (4), Maximum digits in DTMF (9), DTMF dialling timeout (10s), Day of deleting GSM inc. group statistics (1), Prefix before DISA dial-in, CLIP ('-' removes one digit), and Looping of voice message (Off). The 'Send CLIP from GSM/UMTS to VoIP' section shows 'Transfer CLIP from GSM/UMTS' (checked), 'Separating char', and 'Modify ('-' removes one digit)'. A 'Logout' button is visible in the bottom left.

You can define the list of called numbers which will be automatically dialed after DTMF dialing timeout if the customer does not press any button within 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 page. It features a text input field with the value '100' and a note: 'Only 0123456789*#+ characters are allowed'. Below the input field is a list box containing the number '100'. To the right of the list box are three buttons: 'Add', 'Remove', and 'Remove all'. The interface also includes a sidebar and a 'Logout' button.

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

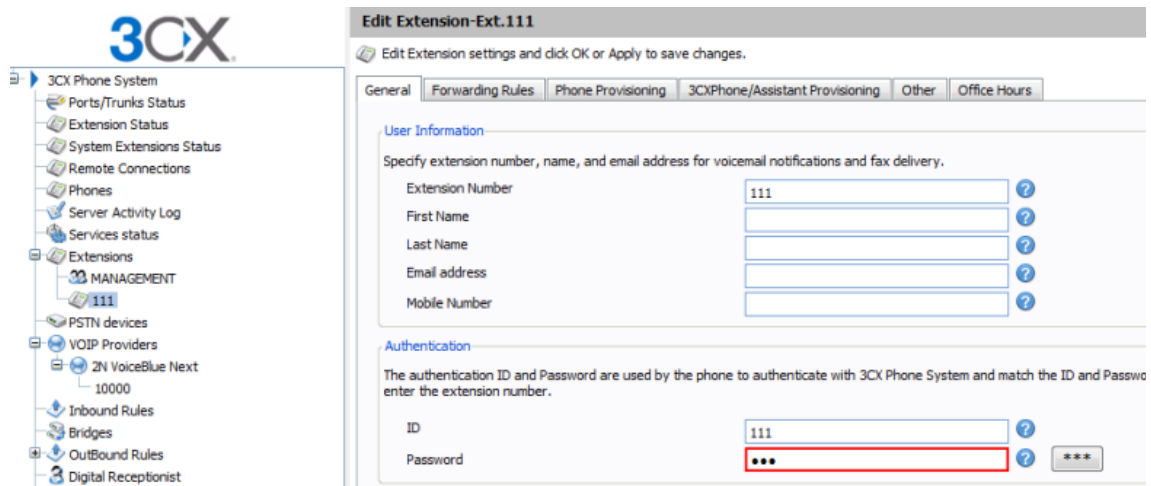
Product	3CXPSDEMO
Version Number	9
Support	n/a
Upgrade insurance	n/a
Number of Simultaneous Calls	2
Number of G729 Channels	0
License key	6IA8-6ZUI-FS7G

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.

- You can check differences between Free version and Commercial Edition in [this document](#) on page 11.

1. Create an Extension

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



The screenshot shows the 3CX Phone System management interface. On the left is a navigation tree with '3CX Phone System' expanded, showing 'Extensions' and '111' selected. The main panel is titled 'Edit Extension-Ext.111' and contains the following sections:

- General** (selected tab):
 - User Information**: Fields for Extension Number (111), First Name, Last Name, Email address, and Mobile Number.
 - Authentication**: Fields for ID (111) and Password (masked with ***).

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	<input type="text" value="10000"/>	
Authentication ID	<input type="text" value="10000"/>	
Authentication Password	<input type="password" value="****"/>	

Simultaneous Calls

Maximum simultaneous calls	<input type="text" value="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 ?
 Connect to Queue / Ring Group ?
 Connect to Digital Receptionist ?
 Voicemail box for Extension ?
 Forward to Outside Number ?
 Send fax to email of extension ?

Same as Out of Office hours

3. Set up Outgoing rules

- a. Prefix called from the extension
- b. Set up the range of extension which can use this trunk (e.g. 111-120)
- c. 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 ?

Apply this rule to these calls
 Define to which outbound calls the rule must apply

Calls to numbers starting with (Prefix) ?
 Calls from extension(s) ?
 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
Route 1	<input type="text" value="2N VoiceBlue Next"/>	<input type="text" value="1"/>	<input type="text"/>
Route 2	<input type="text"/>	<input type="text" value="1"/>	<input type="text"/>
Route 3	<input type="text"/>	<input type="text" value="1"/>	<input type="text"/>

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	Assigned Codecs
Speex	G.711 U-law
iLBC	G.711 A-law
G729	GSM-FR

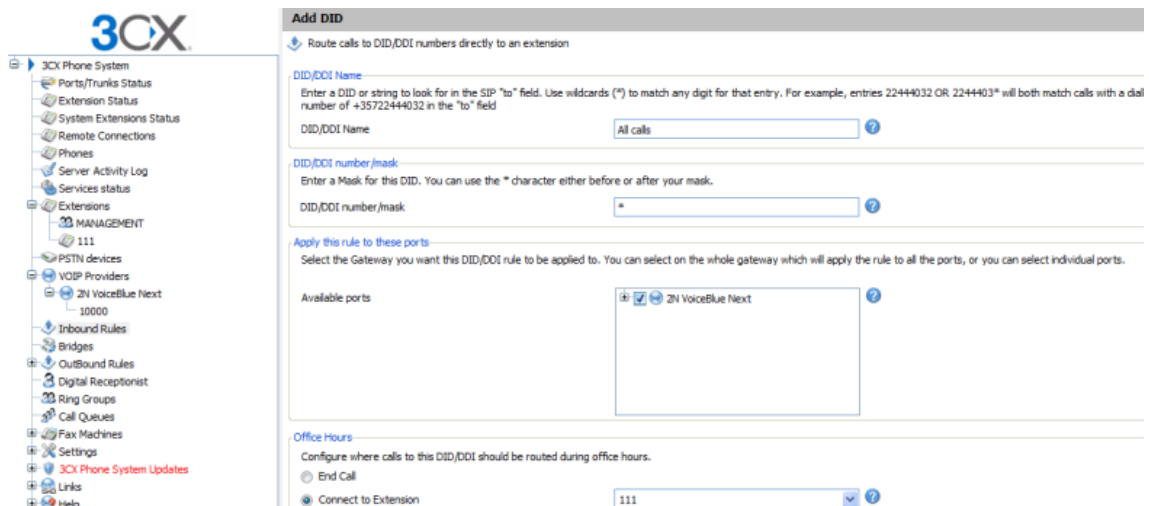
Buttons: Add >, < Remove, Up, Down

4. Create inbound rules

specify numbers or range numbers that 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 voice mail where the call will be connected



3CX

Add DID
 Route calls to DID/DDI numbers directly to an extension

DID/DDI Name: ?

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 ?

5. 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.

More product information:

[2N[®] VoiceBlue Next \(Official Website 2N\)](#)