



2N®

VoiceBlue Next



2N® VoiceBlue Next & Asterisk

connected via SIP trunk

Quick guide

Version 3.00

www.2n.cz

2N® VoiceBlue Next has these parameters:

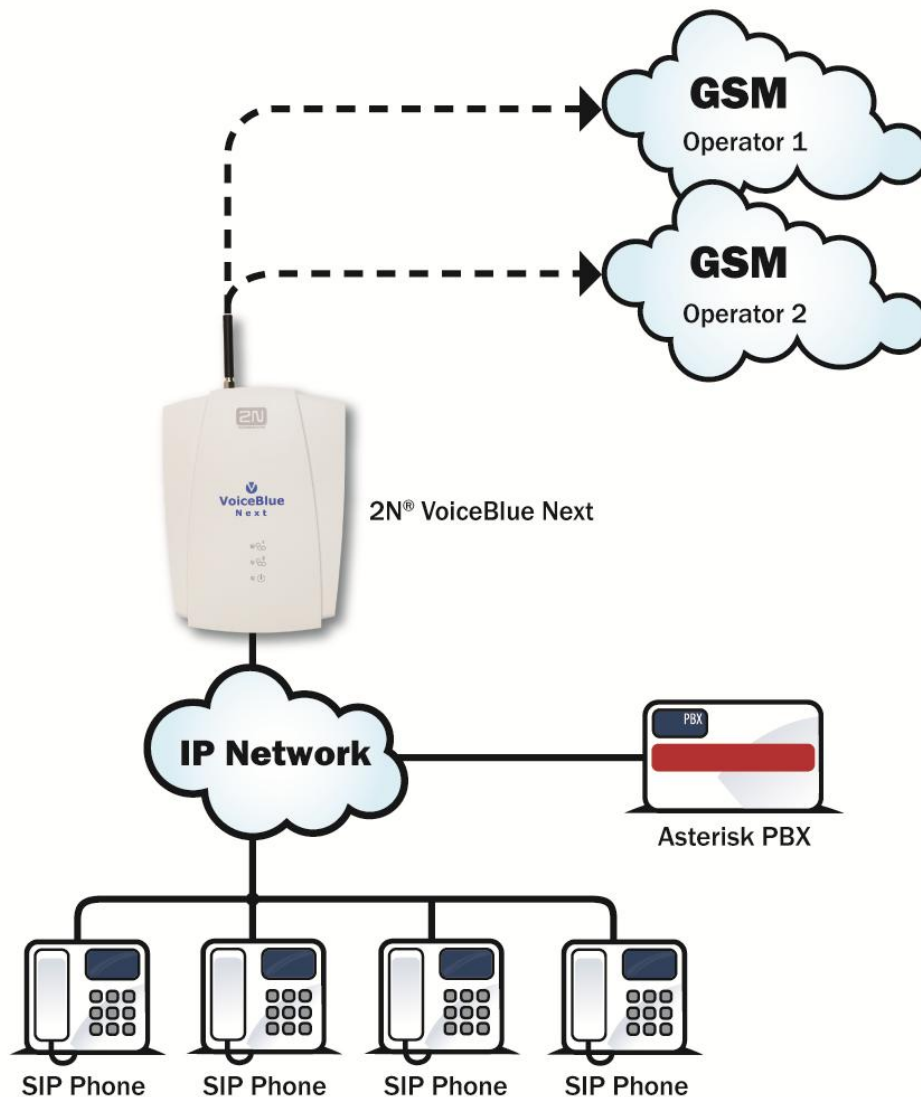
- IP address 10.0.0.20
- Incoming port: 5060

Asterisk parameters:

- IP address 10.0.0.10
- Incoming port: 5060

Scenario

If we have an IP network in which an Asterisk PBX, several SIP phones and 2N® VoiceBlue Next are connected, the configuration would be as shown in the figure below. Furthermore, suppose that the network is addressed as shown in the figure and GSM numbers are all numbers starting with 6, 7 and containing 9 digits.



SIP TRUNK INTERCONNECTION

- 1) For the setting of the trunk between the VoiceBlue Next and your Asterisk PBX, you need to configure “SIP proxy (GSM→IP)” for GSM incoming calls. “SIP proxy (IP→GSM)” is designed only for secure communication with the traffic from your Asterisk. You can specify the IP address and port where the IP packets will be accepted.

If you leave there “0.0.0.0”, the traffic will be unsecured.

To enable incoming calls to Asterisk, you can register the 2N® VoiceBlue Next directly into the Asterisk system. You can register it as “Friend” types in case you require registration on based on username and password or “peer” type (on based of IP address and port).

- SIP registrar...an Asterisk IP address which registers the gateway
- Registration domain – IP address where the gateway is going to be registered
- Username...username under which the gateway shall be registered
- Password...registration password

The screenshot displays the 'Gateway' configuration page for a 2N VoiceBlue Next device. The interface includes a sidebar with navigation options and a main configuration area. Two callouts provide context for the 'SIP proxy (IP→GSM)' settings.

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

Gateway | Update | Restart

Codec priority

Priority 1: G711a (8)
Priority 2: G729 (18)
Priority 3: -----

IP addresses

Setting	Value	Port	Action
SIP proxy (IP→GSM):	10.0.0.10	5060	Set default port
SIP proxy (GSM→IP):	10.0.0.10	5060	Set default port
SIP registrar:	10.0.0.10	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		

SIP registration

Registration expires [s]:	600
Reattempt registration [s]:	10
Registration domain (realm):	10.0.0.10
Caller ID:	100
Username:	100 (Write only)
Password:	100 (Write only)

Callout 1: The IP address where the traffic is sent

Callout 2: The IP address and port which the traffic will come from


2) Configuration of the LCR (Least Cost Routing)

You have to specify prefixes for the operators in the country you are currently located. An example of this would be that in Czech Republic prefix 6 and 7 have a 9 digits number.

The setting is displayed below.

The screenshot displays the 2N Gateway configuration web interface. On the left is a sidebar with the 2N logo and a menu for 'Gateway control' and 'Gateway configuration'. The 'Gateway configuration' menu includes 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 'Prefixes' option is selected. The main content area is titled 'Prefixes' and contains a 'GSM prefix lists' section with tabs for 'Prefixlist 1' through 'Prefixlist 8'. Below this is the 'Basic settings' section, which includes 'GSM network ID' (a text input field) and 'Default count of digits' (a dropdown menu currently set to '9'). There are two tables: 'Table of replaced prefixes' and 'Table of accepted prefixes'. Both tables have a warning: 'Only 0123456789*# characters are allowed'. The 'Table of replaced prefixes' is currently empty. The 'Table of accepted prefixes' contains two entries: '6' and '7'. To the right of each table is a form with fields for 'Prefix', 'Replace with', and '[Digits count]', along with 'Add', 'Remove', and 'Remove all' buttons. At the bottom left is a 'Logout' button with an information icon. At the bottom right are three icons representing different file types or actions.

- 3) You need to create specific guidelines connecting prefixes with the GSM group. In the “GSM group” you will specify settings for SIM cards assigned to this specific group. In the “GSM group assignment” you can assign the module for the appropriate GSM outgoing group.

**Gateway**
Gateway | Update | Restart

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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:

- 0. module
- 1. module

Outgoing:

- 1. Group
- 2. Group

Incoming:

- 1. Group
- 1. Group

4) Configuration of GSM outgoing groups:

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

2N TELECOMMUNICATIONS

Gateway | Update | Restart

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Logout

GSM outgoing groups

1. GSM group | 2. GSM group

General settings

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: ☒

Call length counting: Seconds

Disconnect call

SIM limit exceeded: ☐

Time limit exceeded: ☐

No ALERTING before CONNECT: ☐

Send CLIP from VoIP to GSM/UMTS

Attention! Must be supported by your GSM / UMTS operator. In other case outgoing calls to GSM / UMTS can be rejected!

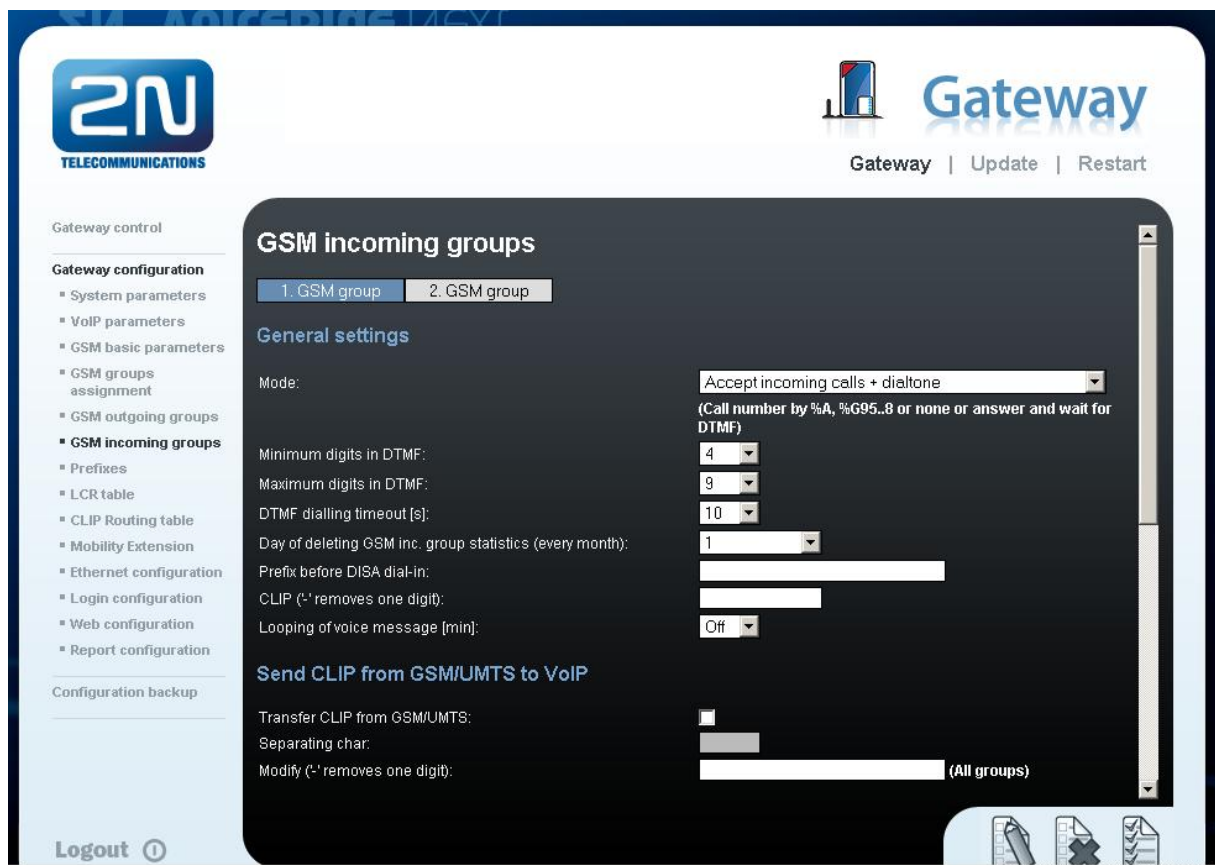
5) Incoming calls

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

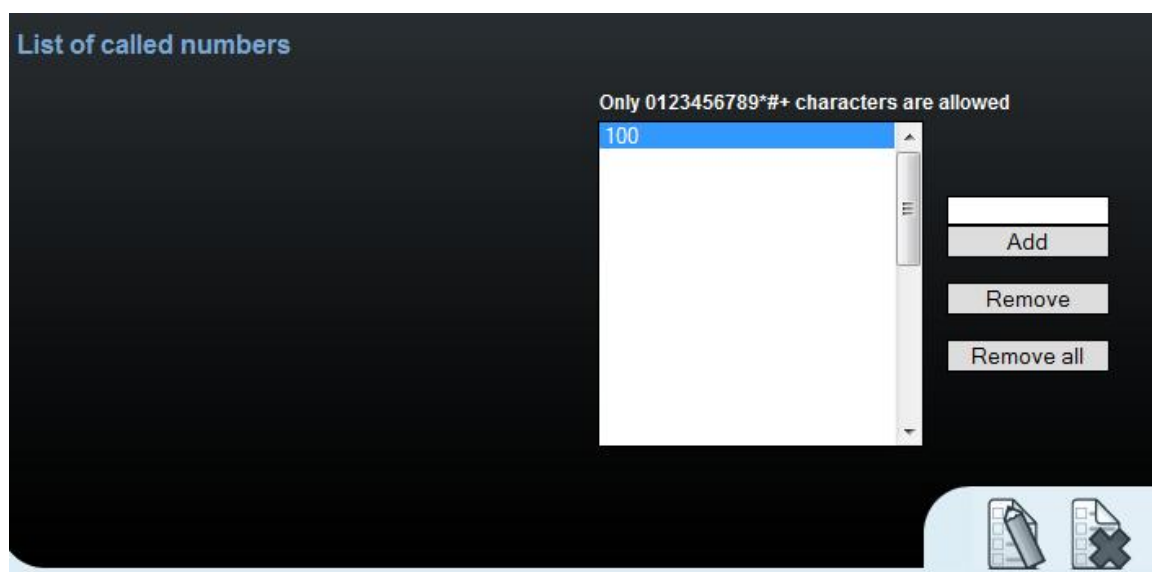
GSM groups assignment

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

In GSM incoming groups you can specify the traits for each GSM incoming group. Choose the mode to Reject, Ignore, Accept incoming calls or Callback.



You can define the list of numbers called. The number will be automatically dialed after the DTMF dialing has timed out. This happens when the customer doesn't press any button until the specific time. At this point, the number will be routed to the extension 100 to your Asterisk (if you set up SIP proxy (GSM->IP) in VoIP parameters).



ASTERISK SETTING

Now add a few lines into the Asterisk configuration for proper routing of outgoing calls to the 2N® VoiceBlue Next gateway and receiving calls coming from the GSM gateway to Asterisk.

1) Outgoing calls

The core of Asterisk connection is saved in the */etc/asterisk/extensions.conf* file.

Open this file in your favorite editor and add the following lines:

```
exten=>_6XXXXXXX,1,Dial(SIP/${EXTEN:0}@10.0.0.20,,r)
```

```
exten=>_7XXXXXXX,1,Dial(SIP/${EXTEN:0}@10.0.0.20,,r)
```

Once you have saved and closed the file, restart Asterisk. From this point forward, all calls starting with 6 and 7 should be routed to the 2N® VoiceBlue Next gateway.

2) Incoming calls

It is highly recommended to make a little restrictions for incoming calls to prevent unauthorized people from calling over your system.

Since the 2N® VoiceBlue Next system works with the SIP, modify the */etc/asterisk/sip.conf* file where the 2N® VoiceBlue Next section could look as follows:

```
[general]
port = 5060
bindaddr = 0.0.0.0
allowgsm=no
context = sip
disallow=all
allow=ulaw

[VoiceBlueNext]
type=peer
host=10.0.0.20
username=voiceblue
secret=password
fromdomain=10.0.0.20
```

Again, restart the Asterisk after saving the file. Then the Asterisk will be ready to receive calls coming from the 2N® VoiceBlue Next gateway.

What to do in case of trouble:

First of all, check our webpage faq.2n.cz and try to see if there is a solution to your problem. In case, you cannot find the proper answer, use the link: How to report an issue on the 2N® VoiceBlue Next.

Here is the direct link:

<https://jira.2n.cz/confluence/pages/viewpage.action?pageId=22513331>



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