



2N®

VoiceBlue Next



2N® VoiceBlue Next & Elastix PBX

connected via SIP trunk

Quick guide

Version 1.00

www.2n.cz

2N® VoiceBlue Next has these parameters:

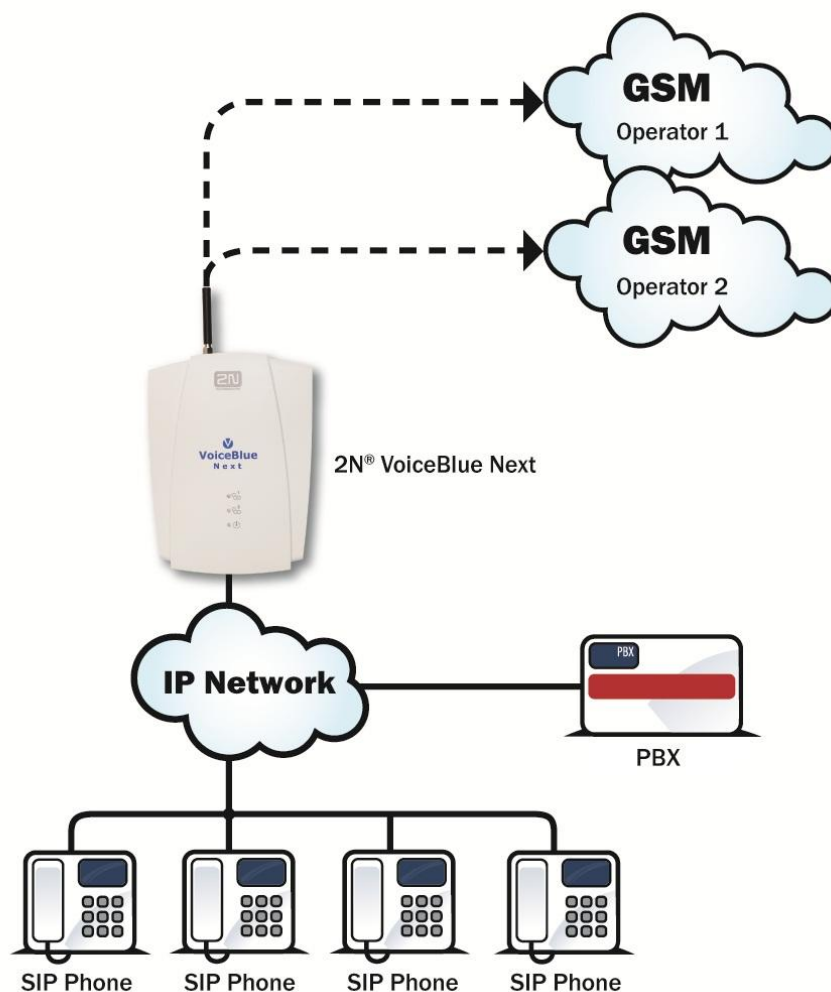
- IP address 192.168.50.45
- Incoming port: 5060
- Firmware: 01.00.04

Elastix PBX:

- IP address 192.168.50.115
- Incoming port: 5060
- Firmware Elastix: 2.0.0
- Firmware Asterisk: 1.6.2.13

Scenario

If we have an IP network in which an Elastix 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 Elastix 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 Elastix PBX. You can specify the IP address and port where the IP packets will be accepted.

The screenshot shows the 'SIP proxy (GSM→IP)' configuration page. The left sidebar lists various configuration categories, with 'VoIP parameters' selected. The main area is divided into sections: 'Codec priority' (with three priority dropdowns set to G711a (8), G711u (0), and G729 (18)), 'IP addresses' (with four rows for SIP proxy, SIP registrar, NAT firewall, and STUN server, each with an IP input field and a 'Set default port' button), and 'Tones generated to VoIP' (with a 'Ring tone to VoIP' dropdown set to 'From GSM'). Two callout bubbles are present: one pointing to the 'SIP proxy (GSM→IP)' IP field with the text 'The IP address where the traffic is sent', and another pointing to the 'Set default port' button with the text 'The IP address and port which the traffic will come from'.

Section	Parameter	Value	Action
IP addresses	SIP proxy (IP→GSM):	0.0.0.0	Set default port
	SIP proxy (GSM→IP):	192.168.50.115	Set default port
	SIP registrar:	0.0.0.0	Set default port
	NAT firewall:	0.0.0.0	Set default port
STUN server	STUN server:	0.0.0.0	Set default port
	Next STUN server request (60-6553, 0=off) [s]:	600	

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.

- 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

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

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

Configuration backup

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)

The screenshot shows the 'Gateway' configuration interface. On the left is a sidebar with a '2N TELECOMMUNICATIONS' logo and a 'Gateway control' menu. The menu includes 'Gateway configuration' (with sub-items: 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 area is titled 'GSM outgoing groups' and has two tabs: '1. GSM group' (selected) 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:'. Under 'Send CLIP from VoIP to GSM/UMTS', 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!'. At the bottom left is a 'Logout' button. At the bottom right are icons for a printer, a document, and a list.

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.

The screenshot shows the 'GSM groups assignment' configuration 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 rectangle.

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.

Gateway | Update | Restart

Gateway control

Gateway configuration

- System parameters
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- GSM groups assignment
- GSM outgoing groups
- GSM incoming groups**
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 - LCR table
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 - Login configuration
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 - Report configuration

Configuration backup

Logout ⓘ

GSM incoming groups

1. GSM group
2. GSM group

General settings

Mode:

Accept incoming calls + dialtone

(Call number by %A, %G95..8 or none or answer and wait for DTMF)

Minimum digits in DTMF:

4

Maximum digits in DTMF:

9

DTMF dialling timeout [s]:

10

Day of deleting GSM inc. group statistics (every month):

1

Prefix before DISA dial-in:

CLIP ('-' removes one digit):

Looping of voice message [min]:

Off

Send CLIP from GSM/UMTS to VoIP

Transfer CLIP from GSM/UMTS:

☐

Separating char:

Modify ('-' removes one digit):

(All groups)

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

List of called numbers

Only 0123456789*#+ characters are allowed

100

Add

Remove

Remove all

ELASTIX PBX SETTING

1) Create an extension

In the “PBX Configuration” and “Extensions” you create an extension as in the print screens below.

Extension: 102

 Delete Extension 102

Used as Destination by 1 Object:

 Add Follow Me Settings

Add Gabcast Settings

Edit Extension

Display Name	<input type="text" value="Ext102"/>
CID Num Alias	<input type="text"/>
SIP Alias	<input type="text"/>

Extension Options

Outbound CID	<input type="text"/>
Ring Time	<input type="text" value="Default"/>
Call Waiting	<input type="text" value="Disable"/>
Call Screening	<input type="text" value="Disable"/>
Pinless Dialing	<input type="text" value="Disable"/>
Emergency CID	<input type="text"/>

Add Extension

Ext102 <102>

Ext103 <103>


This device uses sip technology.

secret	<input type="text" value="102"/>
dtmfmode	<input type="text" value="rfc2833"/>
canreinvite	<input type="text" value="no"/>
context	<input type="text" value="from-internal"/>
host	<input type="text" value="dynamic"/>
type	<input type="text" value="friend"/>
nat	<input type="text" value="yes"/>
port	<input type="text" value="5060"/>
qualify	<input type="text" value="yes"/>
callgroup	<input type="text"/>
pickupgroup	<input type="text"/>
disallow	<input type="text"/>
allow	<input type="text"/>
dial	<input type="text" value="SIP/102"/>
accountcode	<input type="text"/>
mailbox	<input type="text" value="102@device"/>
deny	<input type="text" value="0.0.0.0/0.0.0.0"/>
permit	<input type="text" value="0.0.0.0/0.0.0.0"/>

You have to define type as “friend” and listening port, e.g. 5060 as in the example.

2) Set up the route

Add new route in the section “Outbound Routes”. In the example, the route is called VoiceBlue. If you set up Dial Patterns “0|.” it means that you have to make outbound call via prefix 0.

 Delete Route VoiceBlue

Route Name:

VoiceBlue

Rename

Route CID:

☐ Override Extension CID

Route Password:

PIN Set:

None

Emergency Dialing:

☐

Intra Company Route:

☐

Music On Hold?

default

Dial Patterns

0|.

Clean & Remove duplicates

Dial patterns wizards:

(pick one)

Trunk Sequence

0

SIP/VoiceBlue Next

Add

Submit Changes

3) Set up the trunk

Fill up the „Trunk description“, „Outbound Caller ID“ for the outbound identification.

You can limit maximum VoIP channels via dedicated trunk in the menu. Also if you want to send SIP OPTION packets command regularly to check that the device is still online, turn on the parameter „qualify“ as yes.

General Settings

Trunk Description:	<input type="text" value="VoiceBlue Next"/>
Outbound Caller ID:	<input type="text" value="110"/>
CID Options:	<input type="text" value="Allow Any CID"/> ▼
Maximum Channels:	<input type="text" value="2"/>
Disable Trunk:	<input type="checkbox"/> Disable
Monitor Trunk Failures:	<input type="checkbox"/> Enable

Outgoing Dial Rules

Dial Rules:	<div><div></div><div>Clean & Remove duplicates</div></div>
Dial Rules Wizards:	<input type="text" value="(pick one)"/> ▼
Outbound Dial Prefix:	<input type="text"/>

Outgoing Settings

Trunk Name:	<input type="text" value="VoiceBlue Next"/>
PEER Details:	<div><pre>host=192.168.50.45 type=peer qualify=yes</pre></div>

4) Incoming calls

Incoming calls you can route to the IVR. You find the setting in the section „IVR“.

The screenshot shows the Elastix PBX configuration interface. The top navigation bar includes links for System, Agenda, Email, Fax, PBX, IM, Reports, Extras, and Addons. The left sidebar lists various configuration options, with 'IVR' selected under the 'PBX Configuration' section. The main content area is titled 'Edit Misc Application' and contains the following fields:

- Description: IVR
- Feature Code: 7575
- Feature Status: Enabled
- Destination: (empty)
- Phonebook Directory: Phonebook Directory
- Terminate Call: Hangup
- Extensions: <102> Ext102
- Voicemail: <102> Ext102 (busy)
- IVR: Unnamed

Buttons for 'Submit Changes' and 'Delete' are at the bottom. To the right, there are buttons for 'Add Misc Application' and 'IVR'.

Below this, the 'Digital Receptionist' section is visible, titled 'Edit Menu Unnamed'. It includes a 'Save' button and a 'Delete Digital Receptionist Unnamed' button. The 'Used as Destination by 1 Object:' section contains the following fields:

- Change Name: Unnamed
- Announcement: None
- Timeout: 10
- Enable Directory: [checked]
- VM Return to IVR: [unchecked]
- Directory Context: default
- Enable Direct Dial: [checked]
- Loop Before t-dest: [unchecked]
- Timeout Message: None
- Loop Before i-dest: [unchecked]
- Invalid Message: None
- Repeat Loops: 2

Buttons for 'Increase Options', 'Save', and 'Decrease Options' are at the bottom. Below this, the 'Return to IVR' section is visible, with a checkbox and a '#' button. The 'Leave blank to remove' section contains the following fields:

- Phonebook Directory: Phonebook Directory
- Terminate Call: Hangup
- Extensions: <102> Ext102
- Voicemail: <102> Ext102 (busy)
- IVR: Unnamed

Buttons for 'Add IVR' and 'Unnamed' are at the bottom right.

In the „General Settings“, you can allow anonymous incoming calls as in the picture below.

Security Settings

Allow Anonymous Inbound SIP Calls?: yes



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