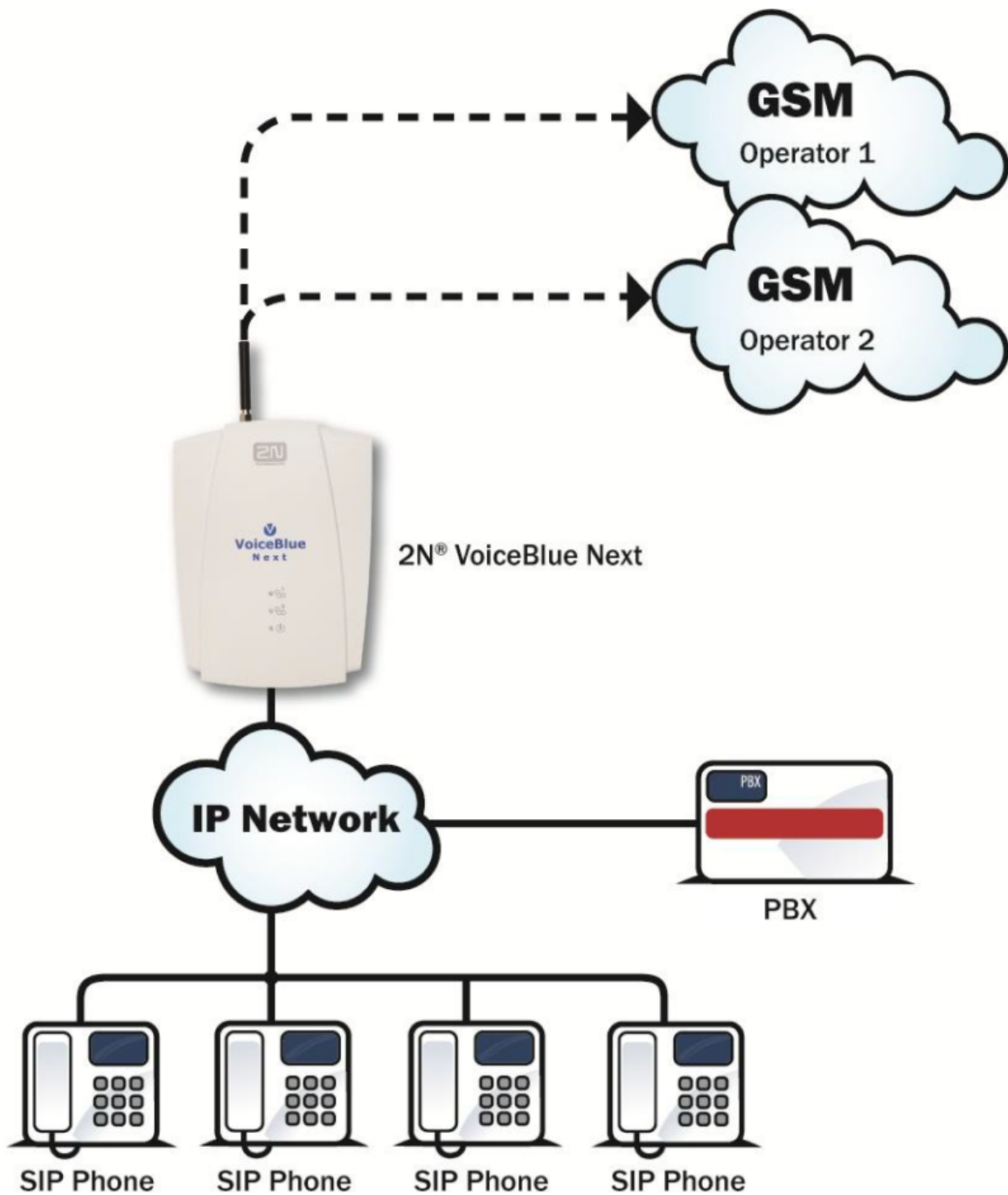


# Elastix PBX - How to interconnect with Elastix PBX?

- 2N<sup>®</sup> 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

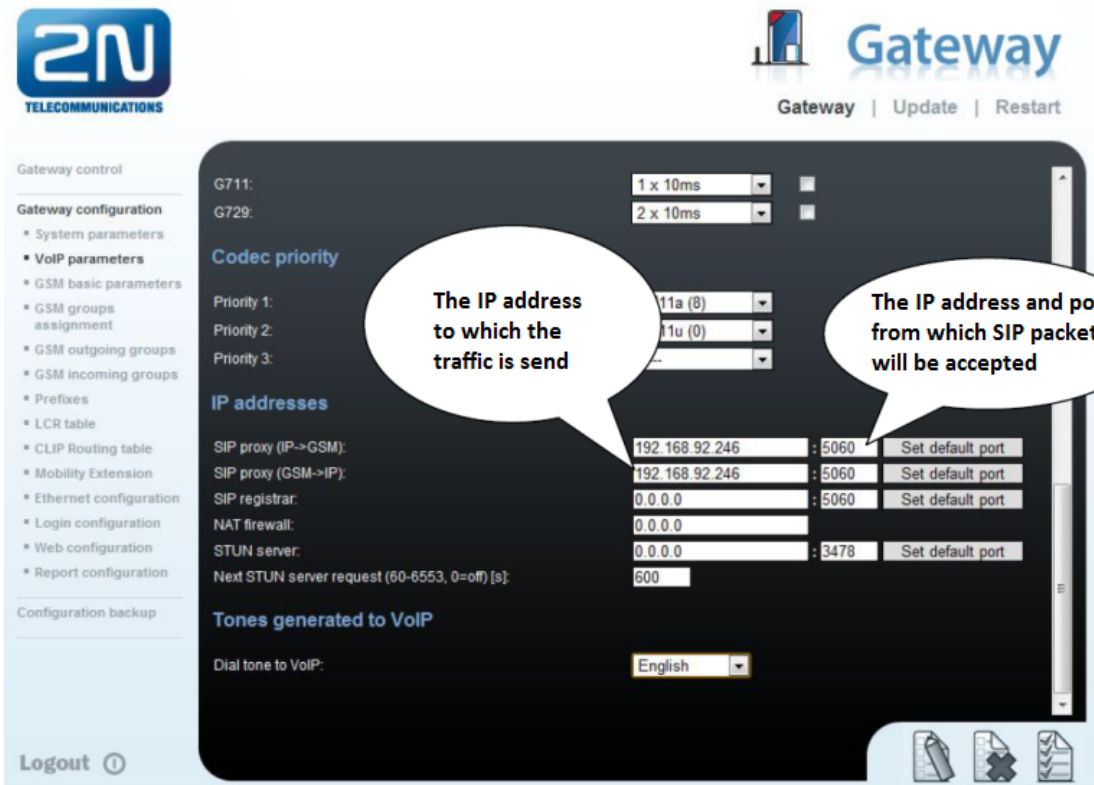
- If we have an IP network in which an Elastix PBX, several SIP phones and 2N<sup>®</sup> VoiceBlue Next are connected, the configuration would be as shown in the figure below. Furthermore, suppose that the network is addressed as shown in the figure and GSM numbers are all numbers starting with 6, 7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



## 2N<sup>®</sup> VoiceBlue Next settings

### 1. SIP trunk interconnection

For the setting of the trunk between the 2N<sup>®</sup> 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 configuration page for a 2N Gateway. The left sidebar contains a navigation menu with categories like 'Gateway control', 'Gateway configuration', and 'Configuration backup'. The main content area is titled 'Gateway' and includes 'Update' and 'Restart' buttons. The configuration is divided into several sections: 'Codec priority' with dropdowns for G711 and G729; 'IP addresses' with fields for SIP proxy (IP->GSM), SIP proxy (GSM->IP), SIP registrar, NAT firewall, and STUN server; and 'Tones generated to VoIP' with a dropdown for 'Dial tone to VoIP'. Two callout boxes are present: one pointing to the 'SIP proxy (IP->GSM)' field with the text 'The IP address to which the traffic is send', and another pointing to the 'SIP proxy (GSM->IP)' field with the text 'The IP address and port from which SIP packets will be accepted'.

## 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 prefixes 6 and 7 have 9-digit numbers. The setting is displayed below.

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.



**2N TELECOMMUNICATIONS** Gateway | Update | Restart

Gateway control

**Gateway configuration**

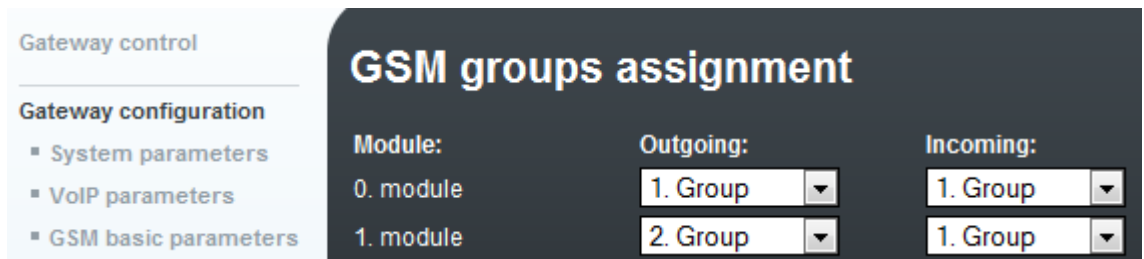
- System parameters
- VoIP parameters
- GSM basic parameters
- GSM groups assignment
- GSM outgoing groups
- GSM incoming groups
- Prefixes
- LCR table**
- CLIP Routing table
- Mobility Extension
- Ethernet configuration
- Login configuration
- Web configuration
- Report configuration

Configuration backup

Logout ⓘ

### 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 control

**Gateway configuration**

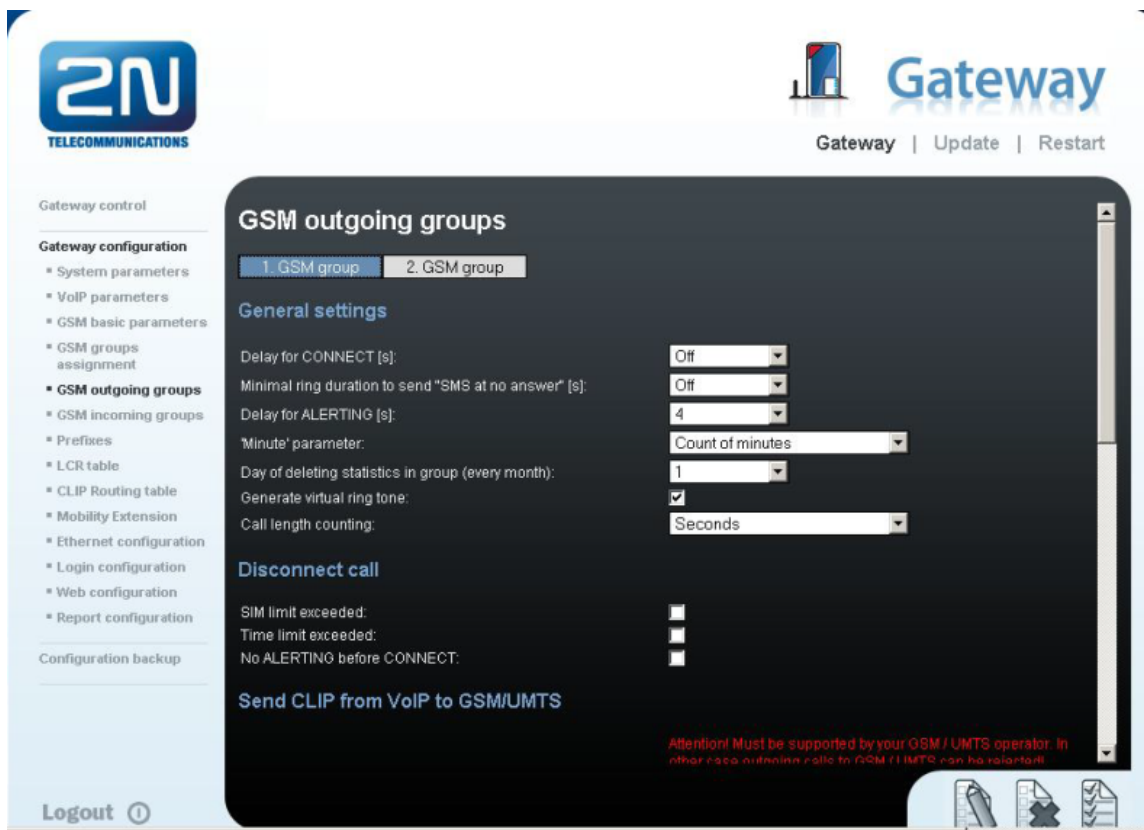
- System parameters
- VoIP parameters
- GSM basic parameters**

### GSM groups assignment

<b>Module:</b>	<b>Outgoing:</b>	<b>Incoming:</b>
0. module	1. Group ▼	1. Group ▼
1. module	2. Group ▼	1. Group ▼

3. 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 page for '2N TELECOMMUNICATIONS'. 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 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:
- Call length counting: Seconds

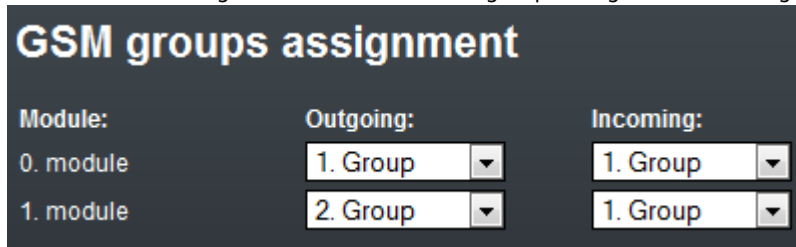
Under 'Disconnect call', there are three checkboxes:

- SIM limit exceeded:
- Time limit exceeded:
- No ALERTING before CONNECT:

At the bottom, there is a section for 'Send CLIP from VoIP to GSM/UMTS' and a red warning: '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 navigation menu with 'GSM outgoing groups' selected.

#### 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 page. It has three columns: 'Module:', 'Outgoing:', and 'Incoming:'. The settings are as follows:

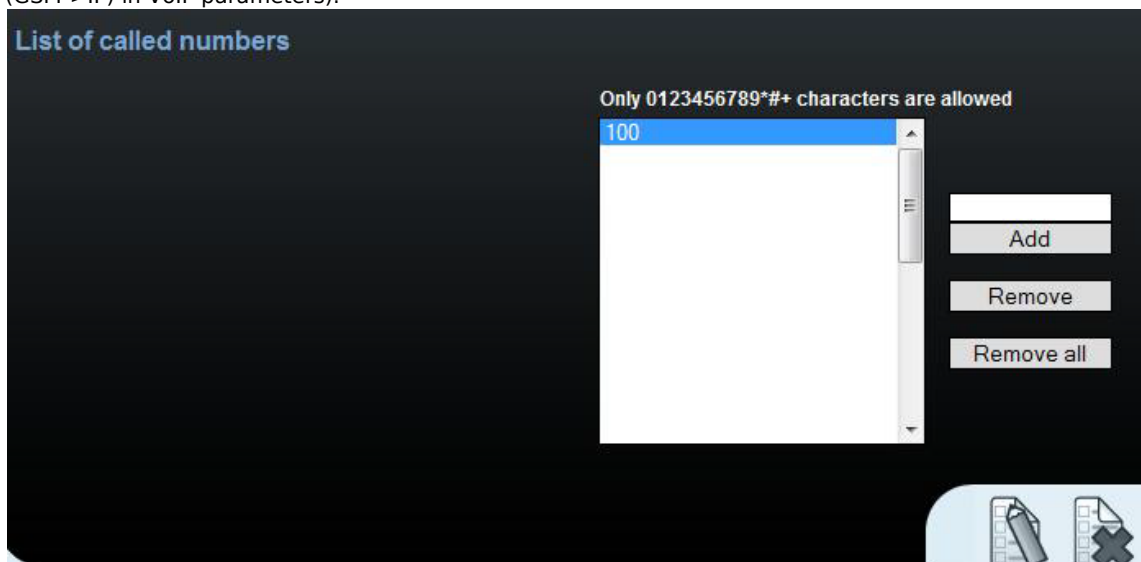
Module:	Outgoing:	Incoming:
0. module	1. Group	1. Group
1. module	2. Group	1. Group

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.



The screenshot shows the 'Gateway' configuration page for '2N TELECOMMUNICATIONS'. 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 [s]: 10; Day of deleting GSM inc. group statistics (every month): 1; Prefix before DISA dial-in: (empty); CLIP (-' removes one digit): (empty); Looping of voice message [min]: Off. There is also a section for 'Send CLIP from GSM/UMTS to VoIP' with fields for 'Transfer CLIP from GSM/UMTS:', 'Separating char:', and 'Modify (-' removes one digit):'. A 'Logout' button is visible in the bottom left corner.

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 does not press button within 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).



The screenshot shows the 'List of called numbers' configuration page. A warning message states: 'Only 0123456789\*#+ characters are allowed'. A list box contains the number '100'. To the right of the list box are three buttons: 'Add', 'Remove', and 'Remove all'. There are also icons for adding and deleting items at the bottom right.

## Elastix PBX settings


1. Create an extension

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

Extension: 102

- Add Extension
- Ext102 <102>
- Ext103 <103>

 Delete Extension 102  
Used as Destination by 1 Object:

 Add Follow Me Settings  
Add Gabcast Settings

Edit Extension

---

Display Name   
 CID Num Alias   
 SIP Alias

Extension Options

---

Outbound CID   
 Ring Time   
 Call Waiting   
 Call Screening   
 Pinless Dialing   
 Emergency CID

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 a 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 an outbound call via prefix 0.

### Delete Route VoiceBlue

Route Name:	VoiceBlue	<input type="button" value="Rename"/>
Route CID:	<input type="text"/>	<input type="checkbox"/> Override Extension CID
Route Password:	<input type="text"/>	
PIN Set:	<input type="text" value="None"/>	
Emergency Dialing:	<input type="checkbox"/>	
Intra Company Route:	<input type="checkbox"/>	
Music On Hold?	<input type="text" value="default"/>	
Dial Patterns	<div style="border: 1px solid #ccc; padding: 5px; min-height: 80px;">0   .</div> <input type="button" value="Clean &amp; Remove duplicates"/>	
Dial patterns wizards:	<input type="text" value="(pick one)"/>	
Trunk Sequence	0 <input type="text" value="SIP/VoiceBlue Next"/> <input type="button" value="🗑️"/> <input type="text"/>	
	<input type="button" value="Add"/>	
<input type="button" value="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 whether the device is still online, turn on the parameter „qualify“ as yes.



### General Settings

Trunk Description:

Outbound Caller ID:

CID Options:

Maximum Channels:

Disable Trunk:  Disable

Monitor Trunk Failures:  Enable

### Outgoing Dial Rules

Dial Rules:

Dial Rules Wizards:

Outbound Dial Prefix:

### Outgoing Settings

Trunk Name:

PEER Details:

```
host=192.168.50.45
type=peer
qualify=yes
```

#### 4. Incoming calls

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

Version \* About us \* Help \* Logout (admin)

System Agenda Email Fax **PBX** IM Reports Extras Addons

PBX Configuration Operator Panel Voicemail Monitoring Endpoint Configurator Conference Batch of Extensions Tools Flash Operator Panel VoIP Provider My Extension

Option  
Unembedded freePBX

Basic  
Extensions  
Feature Codes  
General Settings  
Outbound Routes  
Trunks  
Inbound Call Control  
Inbound Routes  
Zap Channel DIDs  
Announcements  
Blacklist  
CallerID Lookup Sources  
Day/Night Control  
Follow Me  
IVR  
Queue Priorities  
Queues  
Ring Groups  
Time Conditions

### Edit Misc Application

Misc Applications are for adding feature codes that you can dial from internal phones that go to various destinations available in FreePBX. This is in contrast to the **Misc Destinations** module, which is for creating destinations that can be used by other FreePBX modules to dial internal numbers or feature codes.

Edit Misc Application

Description:

Feature Code:

Feature Status:

Destination:

Phonebook Directory:

Terminate Call:

Extensions:

Voicemail:

IVR:

### Digital Receptionist

Edit Menu Unnamed

Used as Destination by 1 Object:

Change Name:

Announcement:

Timeout:

Enable Directory:

VM Return to IVR:

Directory Context:

Enable Direct Dial:

Loop Before t-dest:

Timeout Message:

Loop Before i-dest:

Invalid Message:

Repeat Loops:

Return to IVR

Phonebook Directory:

Terminate Call:

Extensions:

Voicemail:

IVR:

#

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

### Security Settings

Allow Anonymous Inbound SIP Calls?:

More product information:

2N® VoiceBlue Next (Official Website 2N)