



2N[®]

VoiceBlue Next



2N[®] VoiceBlue Next & Panasonic NCP connected via SIP trunk

Quick guide

Version 1.00

www.2n.cz

2N® VoiceBlue Next Parameters

- **IP address:** 192.168.88.2
- **Port:** 5060
- **Firmware:** 03.00.03rc3

PBX Panasonic KX-NCP500VNE

- **IP address:** 192.168.88.101
- **IP DSP:** 192.168.88.102
- **Port:** 5060
- **Port Invite:** 5060
- **Firmware:** 003-000

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 Panasonic PBX. 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.

The screenshot shows the configuration page for a Panasonic PBX. The left sidebar contains a navigation menu with categories like 'Gateway control', 'Gateway configuration', and 'Configuration backup'. The main content area is titled 'SIP proxy (IP->GSM)' and includes the following settings:

Codec	Number of blocks	VAD
G711:	2 x 10ms	<input type="checkbox"/>
G729:	2 x 10ms	<input type="checkbox"/>

Codec priority settings:

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

IP addresses settings:

Field	IP Address	Port	Action
SIP proxy (IP->GSM):	192.168.88.101	5060	Set default port
SIP proxy (GSM->IP):	192.168.88.101	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		

Callout 1: 'The IP address to which the traffic is send' points to the 'SIP proxy (IP->GSM)' field.

Callout 2: 'The IP address and port which will accept traffic from' points to the 'SIP proxy (GSM->IP)' field.

Configuration of the LCR (Least Cost Routing)

The GSM operator has e.g. in our country prefix 7 and 8 with a nine digit the length number. The setting is below.

The screenshot shows the 'Gateway' configuration interface. At the top left is the '2N TELECOMMUNICATIONS' logo. At the top right is the 'Gateway' logo with a mobile phone icon and buttons for 'Gateway | Update | Restart'. The main content area is titled 'Prefixes' and contains 'GSM prefix lists' with tabs for 'Prefixlist 1' through 'Prefixlist 8'. Below this is the 'Basic settings' section with 'GSM network ID' and 'Default count of digits' (set to 9). There are two tables: 'Table of replaced prefixes' and 'Table of accepted prefixes', both with a note 'Only 0123456789*#+ characters are allowed'. Each table has a list of prefixes and associated 'Add', 'Remove', and 'Remove all' buttons. The bottom bar contains a 'Logout' button and three utility icons (edit, delete, refresh).

2) 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 control

Gateway configuration

- System parameters
- VoIP parameters
- GSM basic parameters
- GSM groups assignment
- GSM outgoing groups
- GSM incoming groups
- Prefixes
- LCR table

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

Module:

0. module

1. module

Outgoing:

1. Group

2. Group

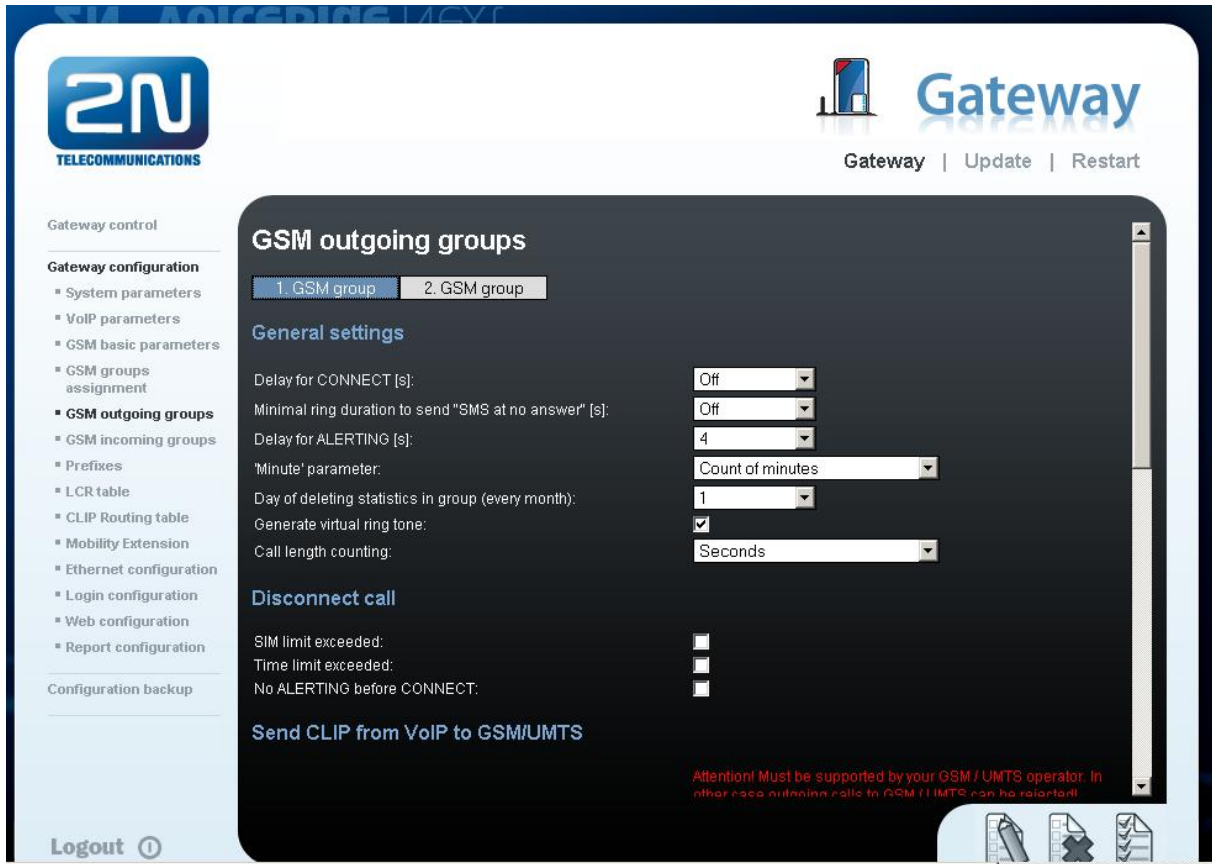
Incoming:

1. Group

1. Group

Configuration of GSM outgoing groups:

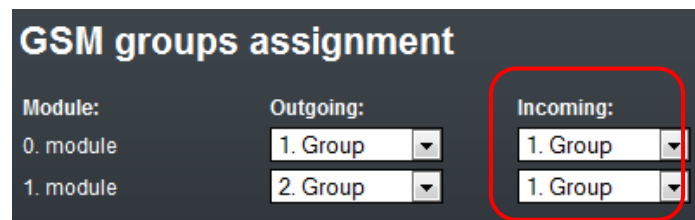
You are able to set up different setting for each GSM group (CLIR, free minutes, Virtual ring tone, roaming and others)



The screenshot shows the 'Gateway' configuration interface 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 configuration options with dropdown menus: '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:'. A section 'Send CLIP from VoIP to GSM/UMTS' is also present. A red warning message at the bottom states: 'Attention! Must be supported by your GSM / UMTS operator. In other case outgoing calls to GSM / UMTS can be refused!'. The interface also features a 'Logout' button and a 'Gateway | Update | Restart' menu.

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 interface. It features a table with columns for 'Module:', 'Outgoing:', and 'Incoming:'. The 'Incoming:' column is highlighted with a red box. The table contains two rows of data:

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 interface. At the top left is the '2N TELECOMMUNICATIONS' logo. At the top right is the 'Gateway' logo with a mobile phone icon and the text 'Gateway | Update | Restart'. The main content area is titled 'GSM incoming groups' and has two tabs: '1. GSM group' (selected) and '2. GSM group'. Under 'General settings', there are several configuration options:

- Mode: Accept incoming calls + dialtone (dropdown)
- (Call number by %A, %G95..8 or none or answer and wait for DTMF)
- Minimum digits in DTMF: 4 (dropdown)
- Maximum digits in DTMF: 9 (dropdown)
- DTMF dialling timeout [s]: 10 (dropdown)
- Day of deleting GSM inc. group statistics (every month): 1 (dropdown)
- Prefix before DISA dial-in: (text input)
- CLIP (' removes one digit): (text input)
- Looping of voice message [min]: Off (dropdown)

 Below this is the 'Send CLIP from GSM/UMTS to VoIP' section:

- Transfer CLIP from GSM/UMTS: (checkbox, unchecked)
- Separating char: (text input)
- Modify (' removes one digit): (text input)

 A note '(All groups)' is visible next to the last input field. On the left side, there is a navigation menu with 'Gateway control' and 'Gateway configuration' sections. The 'Gateway configuration' section includes:

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

 At the bottom left is a 'Logout' button with a help icon. At the bottom right are icons for a pencil, a document with a cross, and a document with a checkmark.

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

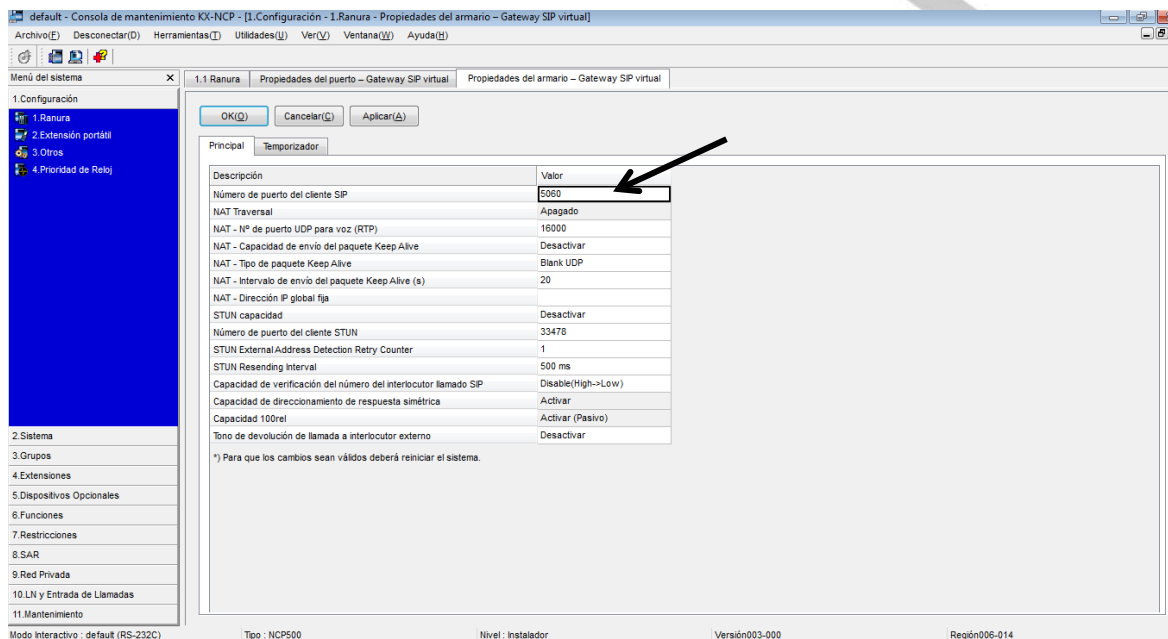
The screenshot shows the 'List of called numbers' configuration interface. At the top, it says 'Only 0123456789*#+ characters are allowed'. Below this is a list box containing the number '100'. To the right of the list box are three buttons: 'Add', 'Remove', and 'Remove all'. At the bottom right are icons for a pencil and a document with a cross.

PANASONIC NCP SETTING (by Masscomm)

1) Connection settings

First of all we need to create a new slot as SIP Trunk (SIP Gateway Virtual type).

By default Panasonic NCP has the port 35060 opened to receive INVITE messages from the other side. We have to switch Virtual rack and V-SIPGW16 card to **mode "OUS"**. Then, we have to change the parameter in the virtual rack and card **Properties of the rack->Virtual SIP Gateway (Trunk) Number of port of SIP Value: 5060**.
(Default value 35060)



After this change the PBX has to be restarted

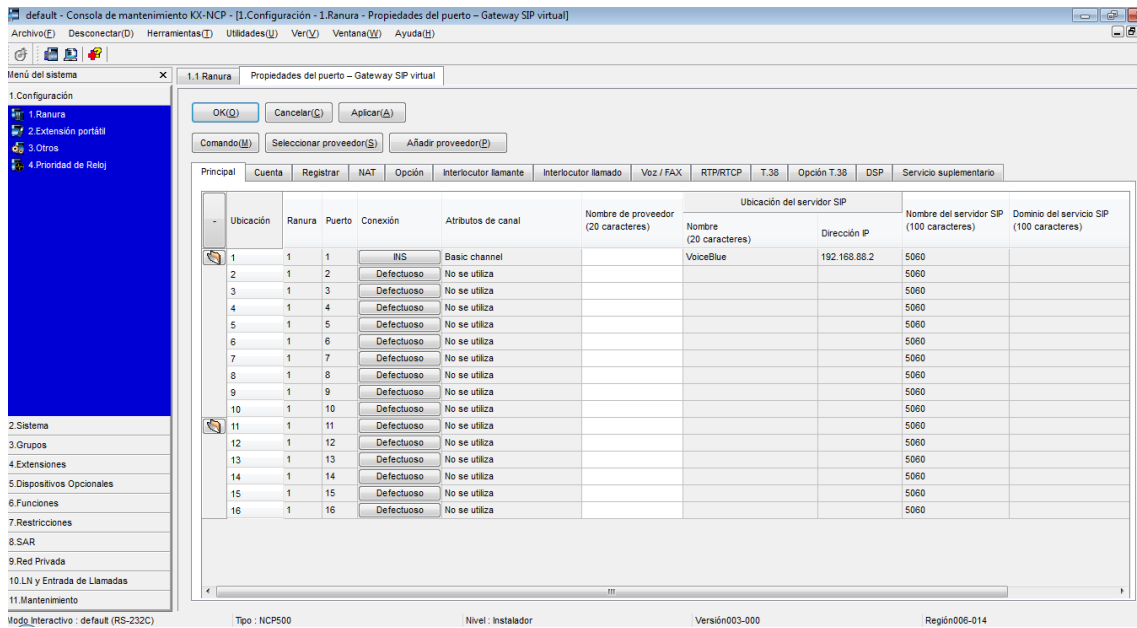
1) SIP Trunk settings

The port is going to be used has to be set as OUS.
In the main tab fill the following params:

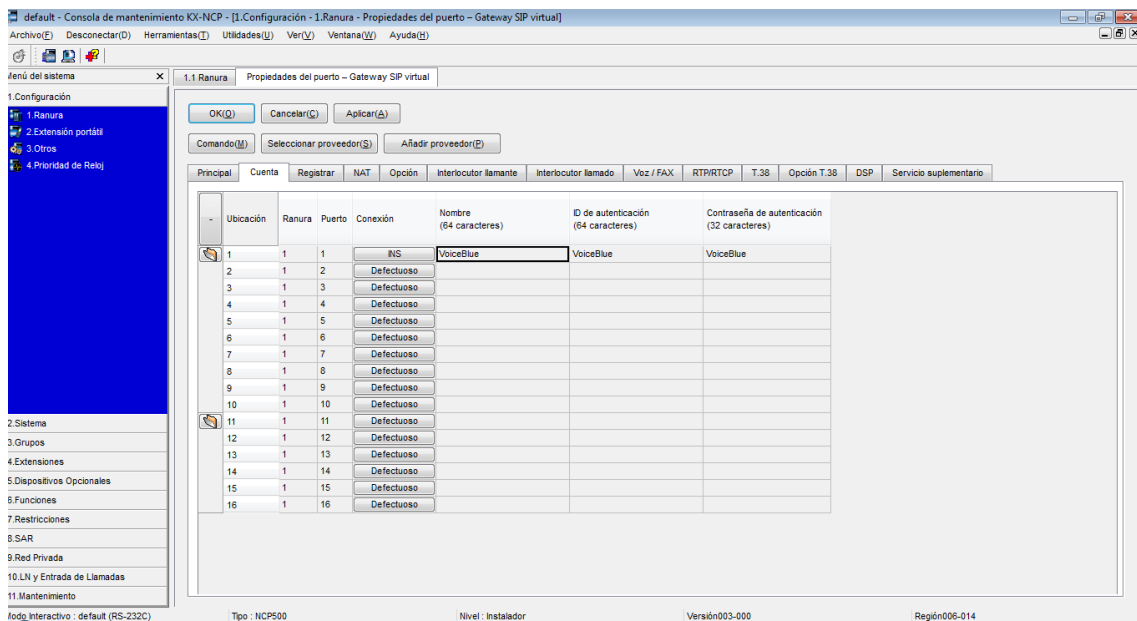
Channel Attribute=Basic Channel;

Name=VoiceBlue

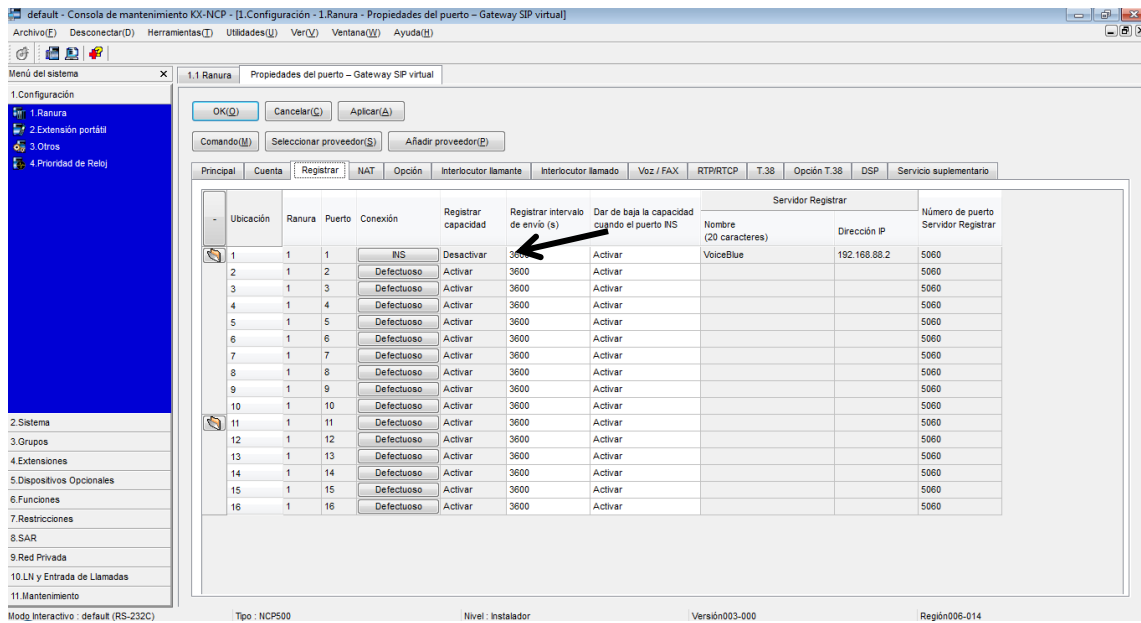
IP address=192.168.88.2 (in the example)



In tab **Account** the fields *Name*, *Authentification ID*, *password*, should be filled. In our case we set all parameters as VoiceBlue



We go to tab **REGISTRAR** and we have to set **REGISTRAR CAPACITY** as **DISABLED**. After that we will configure the server to Registrar Name: **VoiceBlue**; IP= **192.168.88.2**.



After following previous steps the port must be set in mode INS (It should remain marked). We are able to do the testing now of the interconnection of both devices.



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