

Panasonic NCP - How to interconnect with Panasonic NCP PBX?

- 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

2N VoiceBlue 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 just with traffic from your Panasonic PBX. 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 control' interface for a 2N Gateway. The 'IP addresses' section is expanded, showing the following configuration:

Field	Value	Port	Action
SIP proxy (IP→GSM):	192.168.92.246	5060	Set default port
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		

Two callouts are present:

- A callout pointing to the 'SIP proxy (IP→GSM)' row: "The IP address to which the traffic is send"
- A callout pointing to the 'SIP proxy (GSM→IP)' row: "The IP address and port from which SIP packets will be accepted"

2. Configuration of the LCR (Least Cost Routing)

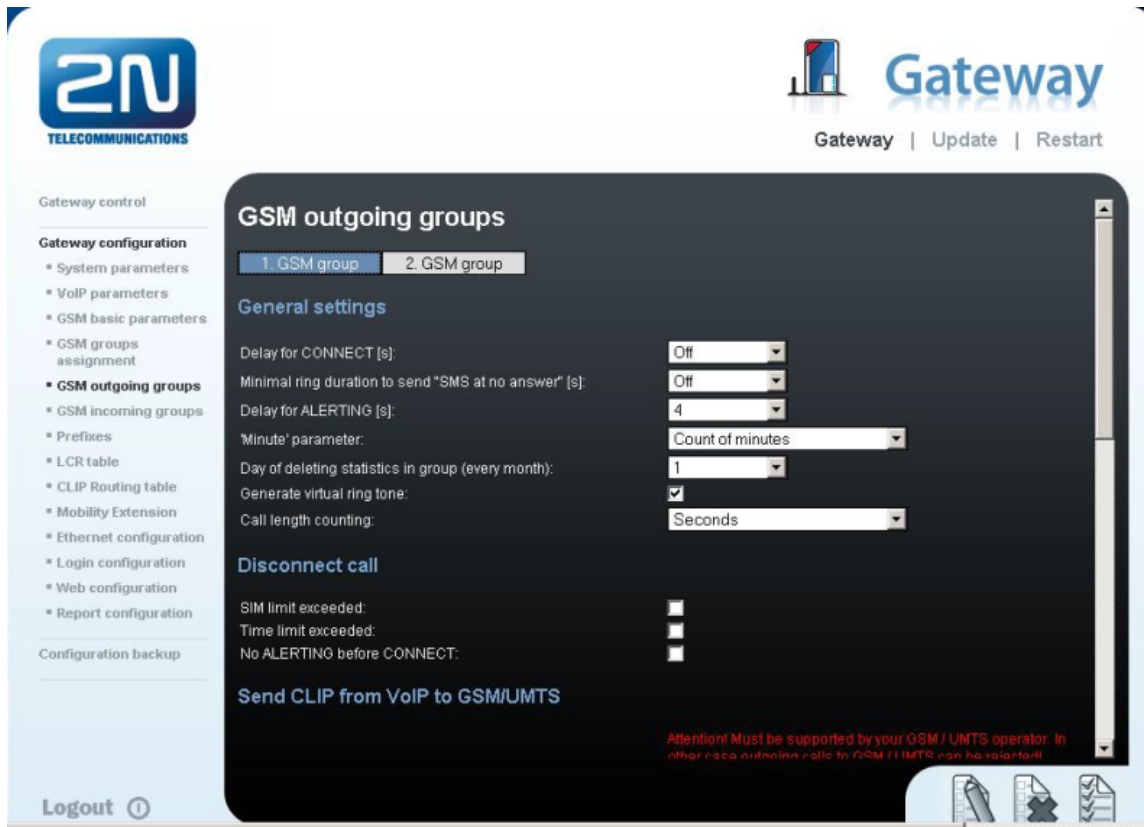
The GSM operator has e.g. in our country prefix 7 and 8 with a 9-digit number. The setting is below.

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.

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

3. 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)



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.

GSM groups assignment

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', there are several fields: 'Mode' (set to 'Accept incoming calls + dialtone'), '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)', and 'Looping of voice message [min]' (Off). Below this is the 'Send CLIP from GSM/UMTS to VoIP' section with fields for 'Transfer CLIP from GSM/UMTS', 'Separating char.', and 'Modify (- removes one digit)'. A 'Logout' button is in the bottom left, and icons for editing and deleting are in the bottom right.

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 Panasonic 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 list box containing the number '100'. Above the list box, a warning message states: 'Only 0123456789*#+ characters are allowed'. To the right of the list box are three buttons: 'Add', 'Remove', and 'Remove all'. Editing and deleting icons are visible in the bottom right corner.

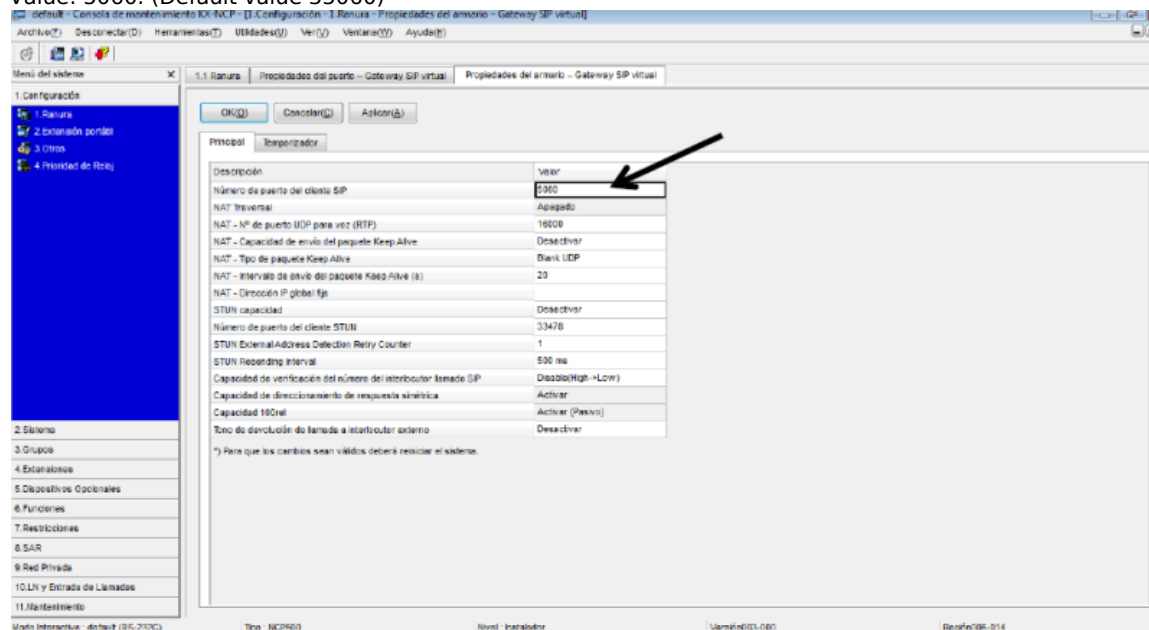
Panasonic NCP settings (by Masscomm)

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)

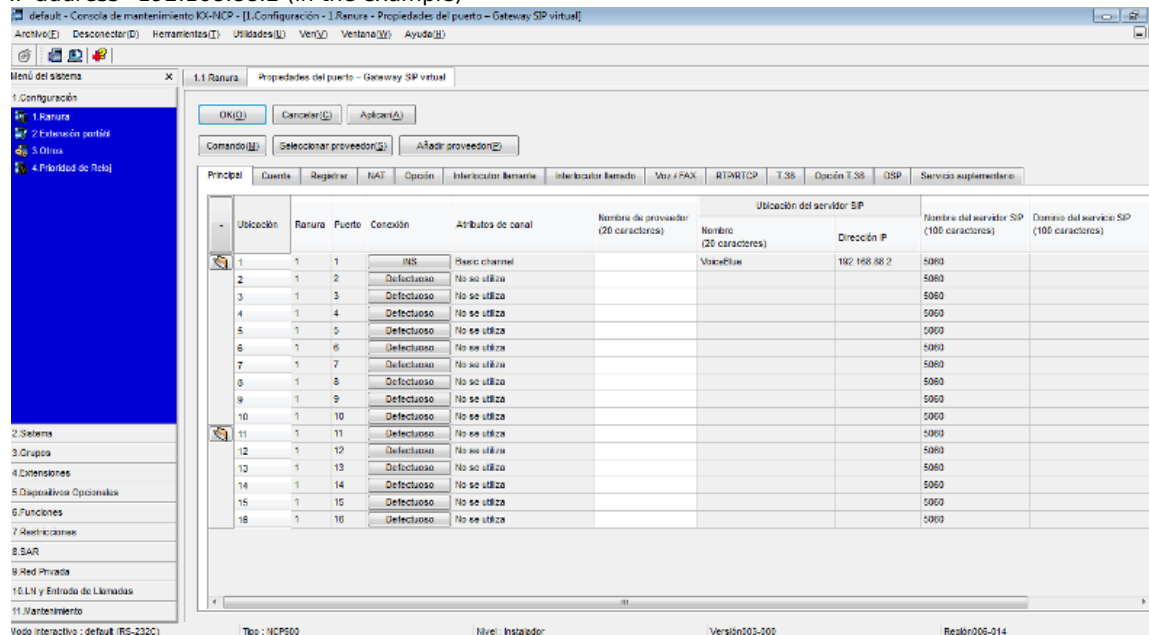


After this change, the PBX must be restarted!

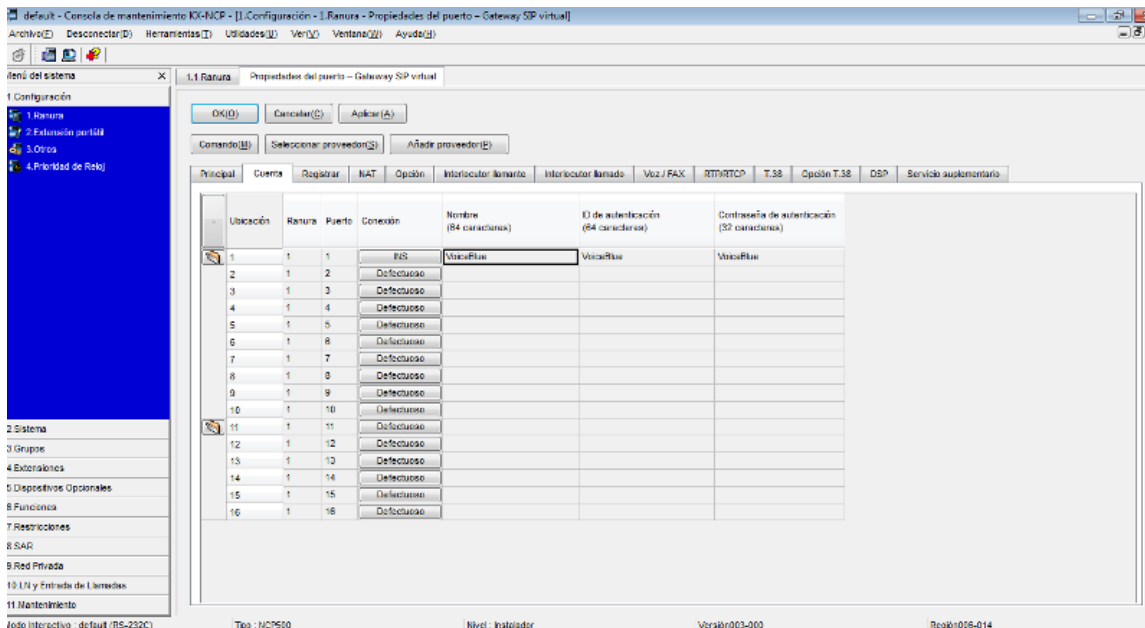
2. SIP Trunk settings

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

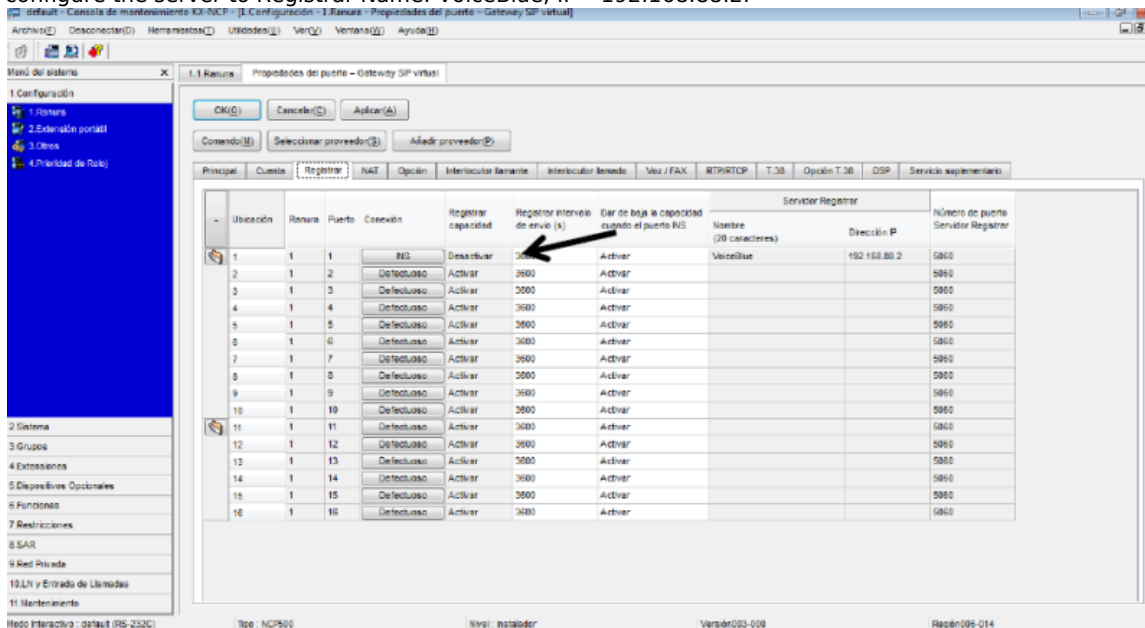
Channel Attribute=Basic Channel
Name=VoiceBlue
IP address=192.168.88.2 (in the example)



In tab Account the fields Name, Authentication 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 having followed all previous steps, the port must be set in mode INS (It should remain marked). Now we are able to do the testing of the interconnection between both devices.

More product information:

2N® VoiceBlue Next (Official Website 2N)