



2N<sup>®</sup>

# VoiceBlue Next



## **2N<sup>®</sup> VoiceBlue Next & Siemens HiPath (series 3000)**

connected via SIP trunk

Quick guide

Version 1.00

[www.2n.cz](http://www.2n.cz)

## 2N® VoiceBlue Next has these parameters:

- IP address 192.168.1.120
- Incoming port: 5060

## Siemens HiPath 3000 parameters:

- IP address 192.168.1.50
- Incoming port: 5060

## 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 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 displays the 2N Gateway configuration web interface. The left sidebar contains a menu with options like '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', and 'Report configuration'. The main content area is titled 'Gateway' and includes buttons for 'Update' and 'Restart'. It features several configuration 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 'Dial tone to VoIP' dropdown. Two callout boxes provide context: one points to the 'SIP proxy (GSM→IP)' field (192.168.1.50) stating 'The IP address to which the traffic is send', and another points to the 'SIP proxy (IP→GSM)' field (0.0.0.0) stating 'The IP address and port which will accept traffic from'. The 'SIP proxy (IP→GSM)' field is set to 0.0.0.0 and port 5060, with a 'Set default port' button next to it.

## 2) Configuration of the LCR (Least Cost Routing)

The GSM operator has e.g. in our country prefix 6 and 7 with a nine digit the length number. The setting is below.

The screenshot displays the 2N Gateway configuration web interface. The top header includes the 2N TELECOMMUNICATIONS logo on the left and the Gateway logo with 'Gateway | Update | Restart' links on the right. A left sidebar menu lists various configuration categories: Gateway control, Gateway configuration (with sub-items 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), and Configuration backup. The main content area is titled 'Prefixes' and contains a 'GSM prefix lists' section with tabs for Prefixlist 1 through 8. Below this is the 'Basic settings' section, which includes fields for 'GSM network ID' and 'Default count of digits' (set to 9). The interface also features two tables: 'Table of replaced prefixes' and 'Table of accepted prefixes'. Both tables have a warning message: 'Only 0123456789\*#+ characters are allowed'. The 'Table of replaced prefixes' currently shows a single entry with a slash (/). The 'Table of accepted prefixes' shows two entries, '6' and '7'. To the right of each table are input fields for 'Prefix' and '[Digits count]', along with 'Add', 'Remove', and 'Remove all' buttons. At the bottom left of the interface is a 'Logout' button, and at the bottom right are three icons representing a document, a document with an 'X', and a document with a checkmark.

- 3) 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 card belongs to which GSM outgoing group.



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

### 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:	Outgoing:	Incoming:
0. module	<div>1. Group</div>	<div>1. Group</div>
1. module	<div>2. Group</div>	<div>1. Group</div>

#### 4) 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). In case you don't have a Ring back tone, set up Delay for ALERTING to option 4.

The screenshot shows the '2N TELECOMMUNICATIONS Gateway' web interface. The left sidebar contains a 'Gateway control' menu with options like 'Gateway configuration', 'System parameters', 'VoIP parameters', 'GSM basic parameters', 'GSM groups assignment', 'GSM outgoing groups', 'GSM incoming groups', 'Prefixes', 'LCR table', 'CLIR Routing table', 'Mobility Extension', 'Ethernet configuration', 'Login configuration', 'Web configuration', and 'Report configuration'. 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 fields: 'Delay for CONNECT [s]' (Off), 'Minimal ring duration to send "SMS at no answer" [s]' (Off), 'Delay for ALERTING [s]' (4, highlighted with a red box), 'Minute' parameter (Count of minutes), 'Day of deleting statistics in group (every month)' (1), 'Generate virtual ring tone' (checked), and 'Call length counting' (Seconds). Below this is a 'Disconnect call' section with three checkboxes: 'SIM limit exceeded', 'Time limit exceeded', and 'No ALERTING before CONNECT'. At the bottom, there is a 'Send CLIP from VoIP to GSM/UMTS' section and a red warning message: 'Attention! Must be supported by your GSM / UMTS operator. In other case outgoing calls to GSM / UMTS can be restarted!'. The bottom left has a 'Logout' button and the bottom right has icons for a phone, a document, and a checkmark.

#### 5) 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' form. It has three columns: 'Module:', 'Outgoing:', and 'Incoming:'. The 'Module:' column has two rows: '0. module' and '1. module'. The 'Outgoing:' column has two dropdown menus: '1. Group' for '0. module' and '2. Group' for '1. module'. The 'Incoming:' column has two dropdown menus: '1. Group' for '0. module' and '1. Group' for '1. module'. The 'Incoming:' column is highlighted with a red box.

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.

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

## 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 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 101 to your PBX (if you set up SIP proxy (GSM->IP) in VoIP parameters).

## List of called numbers

Only 0123456789\*#+ characters are allowed

101

Add

Remove

Remove all

# SIEMENS HiPath 3000 version 8.0

## 1) Create a new IP trunk

Add a new trunk – **Trunks** → **IP Trunks**

Number trunks = number VoIP channels

You need to assign IP trunk to **Trunk group 16** (always for IP trunk). This trunk is called interwork in the picture below.

The screenshot shows the 'IP Trunks' configuration window. At the top, there are tabs for 'Trunks', 'Routes', 'Routing parameters', 'ISDN parameters', 'LCOSS', 'QSIG features', 'IP Trunks', and 'E.164 table'. The 'IP Trunks' tab is selected. Below the tabs, there is a 'Selection' section with 'Gatekeeper HG1500', 'Slot 5', and a checked 'Enable gateway resources' option. The main section is titled 'Trunks' and contains a table with 8 rows. The table has columns for 'Trunk', 'Code', 'Type', and 'Route'. The first two columns are numbered 1 through 8. The 'Type' column for all rows is 'IP Trunking', and the 'Route' column for all rows is 'interwork'. To the right of the table, there is a 'Number' dropdown menu, which is highlighted with a red box and set to 'IP Trunking'. Below this dropdown is an 'Add' button. Further down, there is a 'Selected line...' section with a 'Delete' button. At the bottom, there is a 'Configured line' section showing 'Number 8'.

	Trunk	Code	Type	Route
1	Line 5	7805	SIP Provider 2	SIP 2
2	Line 6	7806	SIP Provider 2	SIP 2
3	Line 7	7807	IP Trunking	interwork
4	Line 8	7808	IP Trunking	interwork
5	Line 9	7809	IP Trunking	interwork
6	Line 10	7810	IP Trunking	interwork
7	Line 11	7811	IP Trunking	interwork
8	Line 12	7812	IP Trunking	interwork

Number: IP Trunking

Add

Selected line...

Delete

Configured line

Number 8

The setting of Trunk group 16 you can find in the picture below.

Important is to use en-block setting for sending of dialed number.

The screenshot shows a PBX configuration window with the following sections:

- Trunks:** Trk Grp 1, Trk Grp 2, Trk Grp 3, Trk Grp 4, Trk Grp 5, Trk Grp 6, Trk Grp 7, Trk Grp 8, Trk Grp 9, Trk Grp10, Trk Grp11, Trk Grp12, SIP 2, Trk Grp14, Trk Grp15, **interwork** (selected).
- Route Name:** Name: interwork
- CO code:** 2nd trunk code ☐
- PABX number-incoming:** Country code, Local area code, PABX number, ☐ Location number current: Trk Grp. 1
- PABX number-outgoing:** Country code, Local area code, PABX number, ☐ Suppress station number
- Route prefix:** 859. Note: With active LCR this field will be used as incoming route prefix. All routes are allowed to have the same prefix.
- Overflow route:** None
- Digit transmission:** en-bloc sending (highlighted with a red box)
- Numbering plan:**
  - Called Party Number:** ☒ System check, ☐ ISDN numbering plan, ☐ Private numbering plan, ☐ Unknown numbering plan
  - All others:** ☒ System check, ☐ ISDN numbering plan, ☐ Private numbering plan, ☐ Unknown numbering plan
- Site:** ☒ System check, ☐ Private network, ☐ Always station
- Switch:**
  - ☒ COLP, ☐ no DIV.LEG-Info, ☐ Intern call like extern, ☒ Notify send, ☐ With sending complete
  - ☐ without CLIP, ☐ Always use DSP, ☐ Without CCNR, ☐ No SETUP ACK.
- Buttons:** Reset, Apply, Help

## 2) LCR SETTING IN PBX

Enter the menu **"Least cost routing" → "Dial plan"**

### **Example setting of Dialed digits:**

0C6Z means: 0... prefix for outgoing calls from PBX

C... user get dial tone (morse A)

6... prefix to GSM network

Z... unlimited number of digits

Now the prefix you have to send to Route table (in our example Route table 3)



Flags and COS | Dial plan | LCR - schedule

Digit analysis wizard

	Name	Dialed digits	Route table	Acc. code	COS	Emergency
1	normal CALL	0CZ	1	No	yes	No
2	SIP call	9CZ	2	No	yes	No
3	VoiceBlue GSM	0C6Z	3	No	yes	No
4	VoiceBlue GSM	0C7Z	3	No	yes	No
5			-	No	yes	No
6			-	No	yes	No
7			-	No	yes	No
8			-	No	yes	No
9			-	No	yes	No
10			-	No	yes	No
11			-	No	yes	No
12			-	No	yes	No
13			-	No	yes	No
14			-	No	yes	No

Route table: 3

Dial rule wizard

Dialing rules table

	Route	Dial rule	min. COS	Schedule	Warning
1	interwork	4 SIP int	15	-	None
2	-	-	15	-	None
3	-	-	15	-	None
4	-	-	15	-	None
5	-	-	15	-	None

Route table 1: Digit-by-digit

Choose your Route table and press “Dial rule wizard”. Now you are able to set up Dial rule format A. It means repeat all digits after C (0 will be stripped from called number).

Dial rule wizard

Edited dial rule: SIP int

Network provider's method of: Main network supplier

Access code:

Pause (max. 12 secs.):

Authorization code:

Dial rule format: A

min. COS: 15

Schedule: -

Warning: None

Type of Number (TON): Unknown

Help OK Cancel

### 3) Setting of VoIP card - via web interface (HG 1500 V8.0)

Firstly, you need to have licenses for VoIP channels (2 channels should be open as a standard)

Enter the menu: **Explorers → Voice Gateway → PBX → Nodes**

**Node 1** needs to be configured for incoming traffic from 2N® VoiceBlue gateway. This setting is for routing to your own system.

LAN trunking protocol needs to be “Native SIP” and IP address is the IP of the Siemens HiPath 300.

The screenshot shows the HG 1500 V8 web interface. On the left is a navigation tree with categories: Front panel, Wizard, Explorers, Maintenance, Help, and Logoff. Under Explorers, the path is Voice Gateway → PBX → Nodes. The main content area is titled 'PBX Node / IP Addresses' and shows configuration for Node Number: 1. The 'LAN trunking protocol' is set to 'Native SIP' (highlighted with a red box). The 'LAN Trunking type' is set to 'Standard Trunking'. The 'HXG Gatekeeper Board 1 - IP Address' is set to '192.168.1.50' (highlighted with a red box). Below this, there are input fields for HXG Board 2 through Board 8, all set to '0.0.0.0'. At the bottom, there is an 'Alive Monitoring' checkbox which is unchecked.

**Node 2** needs to be configured for outgoing traffic to 2N® VoiceBlue gateway.

LAN trunking protocol needs to be “Native SIP” and IP address is the IP of 2N® VoiceBlue gateway (192.168.1.120)

The screenshot shows the HG 1500 V8 web interface for Node Number: 2. The 'LAN trunking protocol' is set to 'Native SIP' (highlighted with a red box). The 'LAN Trunking type' is set to 'Standard Trunking'. The 'HXG Gatekeeper Board 1 - IP Address' is set to '192.168.1.120' (highlighted with a red box). Below this, there are input fields for HXG Board 2 through Board 4, all set to '0.0.0.0'. The 'HXG Board 5' field is not visible in this view.

Siemens HiPath can check the connection with gateway by setting of “Alive monitoring”. You are able to set it up in Node setting and you can choose PING or TCP IP monitoring. Both methods are supporting by 2N® VoiceBlue gateway.

## Routing

Now, you have to set up routing digits to your predefined Nodes.

### Example of setting:

- Number 6 and 7 are routed to the Node 2. There is a gateway 2N® VoiceBlue Next
- Number 1 is routed to the Node 1. This node is for own Siemens HiPath PBX.

Front panel Wizard Explorers Maintenance Help Logoff HG 1500 V8

**Voice Gateway**

- H.323 Parameters
- SIP Parameters
- Codec Parameters
- Internet Telephony Service Provider
- Destination Codec Parameters
- PBX**
  - IP Networking Data
  - Nodes
    - 1
    - 2
  - Routing
    - 1 <1>
    - 6 <2>
    - 7 <2>
- Clients
  - ISDN Classmark

### PBX Route Call Address

Node Number: 2

Station Number:

Service: Voice

## Codec setting

Enter the menu **Voice Gateway → Codec Parameters**

You can set up priorities for codecs.

The Siemens HiPath support DTMF via RFC 2833.

**Voice Gateway**

- H.323 Parameters
- SIP Parameters
- Codec Parameters**
- Internet Telephony Service Provider
- Destination Codec Parameters
- PBX
  - IP Networking Data
  - Nodes
    - 1
    - 2
  - Routing
    - 1 <1>
    - 6 <2>
    - 7 <2>
- Clients
  - System
  - H.323
  - SIP
  - ISDN Classmark

### Codec Parameters

Codec	Priority	Voice Activity Detection	Frame Size
G.711 A-law	<span style="border: 2px solid red; padding: 2px;">Priority 2</span>	VAD: <input type="checkbox"/>	30 msec
G.711 μ-law	<span style="border: 2px solid red; padding: 2px;">Priority 7</span>	VAD: <input type="checkbox"/>	30 msec
G.723	<span style="border: 2px solid red; padding: 2px;">not used</span>	VAD: <input type="checkbox"/>	30 msec
G.729A	<span style="border: 2px solid red; padding: 2px;">Priority 1</span>	VAD: <input type="checkbox"/>	20 msec
G.729AB	<span style="border: 2px solid red; padding: 2px;">not used</span>	VAD: <input checked="" type="checkbox"/>	20 msec

**T.38 Fax**

T.38 Fax: ☒

Use FillBitRemoval: ☒

Max. UDP Datagram Size for T.38 Fax (bytes):

Error Correction Used for T.38 Fax (UDP): t38UDPRedundancy

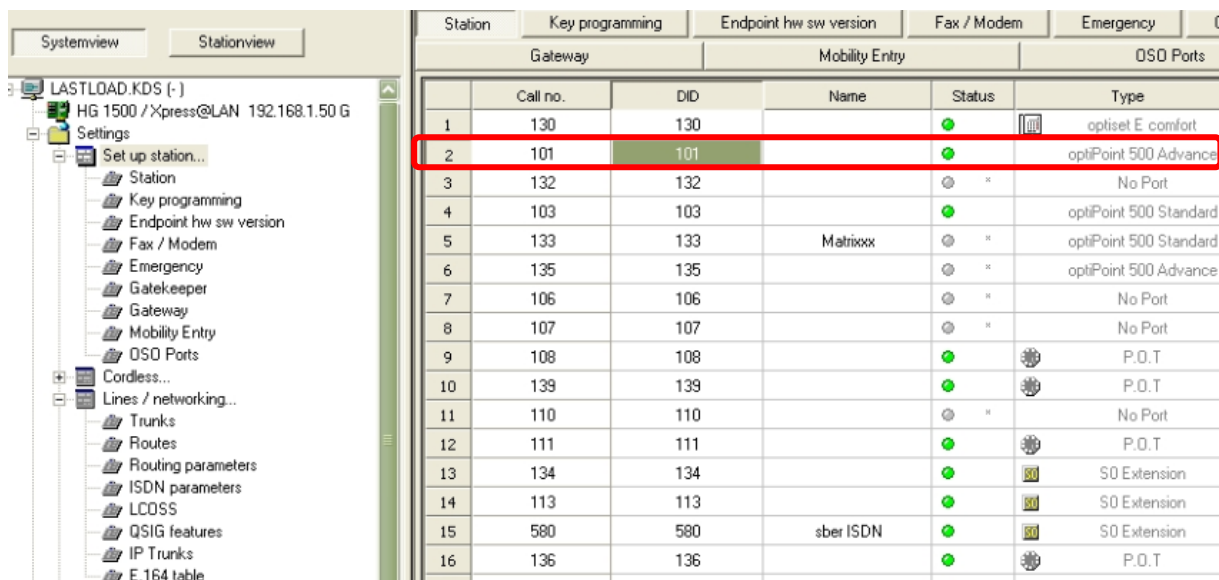
**Misc.**

ClearChannel: ☒      Frame Size: 20 msec

## Incoming call from 2N® VoiceBlue Next

In the VoIP card we already set up routing for prefix “1” to own Siemens HiPath PBX.

Now, the number coming from 2N® VoiceBlue Next is compared with DID number and routed to the particular phone.



Station	Call no.	DID	Name	Status	Type
1	130	130		●	optiset E comfort
2	101	101		●	optiPoint 500 Advance
3	132	132		●	No Port
4	103	103		●	optiPoint 500 Standard
5	133	133	Matrixxx	●	optiPoint 500 Standard
6	135	135		●	optiPoint 500 Advance
7	106	106		●	No Port
8	107	107		●	No Port
9	108	108		●	P.O.T
10	139	139		●	P.O.T
11	110	110		●	No Port
12	111	111		●	P.O.T
13	134	134		●	S0 Extension
14	113	113		●	S0 Extension
15	580	580	sber ISDN	●	S0 Extension
16	136	136		●	P.O.T



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