



2N[®]

VoiceBlue Lite



2N[®] VoiceBlue Lite & Siemens HiPath (series 3000)

connected via SIP trunk

Quick guide

Version 1.00

www.2n.cz

2N® VoiceBlue Lite 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 Lite and your PBX you need to configure SIP proxy (GSM→IP) for GSM incoming calls. The setting is in the VoIP menu. 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 shows a configuration window for SIP settings. It includes fields for IP addresses, checkboxes for 'Use default port', and port numbers. Two callout boxes provide additional context: one points to the 'SIP proxy (GSM->IP)' IP field, and another points to the 'SIP proxy (IP->GSM)' IP and port fields.

Field	Value	Use default port	Port
SIP proxy (IP->GSM)	0.0.0.0	<input checked="" type="checkbox"/>	5060
SIP proxy (GSM-> IP)	192.168.1.50	<input checked="" type="checkbox"/>	5060
SIP registrar :	0.0.0.0	<input checked="" type="checkbox"/>	5060
NAT firewall :	0.0.0.0		
STUN server :	0.0.0.0	<input checked="" type="checkbox"/>	3478
Next STUN server request : (60 - 6553s)	600		[s]

Tones generated to VoIP :
Dial tone to VoIP : Transfer from GSI

Buttons: Save to the gate, Load from the gate, Default, Save to file

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 a configuration window for LCR. On the left, a tree view shows the configuration hierarchy: Gateway control, Gateway configuration (expanded), System parameters, VoIP parameters, GSM basic parameters, GSM groups assignment, GSM outgoing groups, GSM incoming groups, Prefixes, LCR table (selected), CLIP Routing table, and Mobility extension. Below this is a 'Restart' button.

The main configuration area has tabs for 'Prefix list 1' through 'Prefix list 8'. It features two tables:

- Table of replaced prefixes :** Contains a single entry: /
- Table of prefixes :** Contains two entries: 6 and 7

Each table has 'Add', 'Edit', 'Remove', and 'Remove all' buttons. To the right, there are fields for 'GSM network ID' and 'Default count of digits' (set to 9). At the bottom, there are buttons for 'Save to the gate', 'Load from the gate', 'Default', and 'Save to file'.

- 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.

LCR table			
Prefix List	Valid from/to	Outgoing destination	Call duration limit
1/	0:00/24:00	1	0
2/	00:00/24:00	2	0

Add

Edit

Remove

Remove all

Load from the gate

Save to the gate

Save to file

Default

File Gateway Gateway control Settings Help

Topics | Alphabetical glossary

- 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
 - Restart

Groups assignment :

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

Save to the gate Load from the gate

Default Save to file

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.

GSM outgoing groups

1 GSM group | 2 GSM group | 3 GSM group | 4 GSM group

Roaming enabled for network code: Delay for CONNECT [s]:

CLIR: **Delay for ALERTING [s]: (0 = off)**

Max. number of called minutes: (0 = off) Day of deleting stats in group:

SMS messages number: (0 = off) Minimal ring duration to send "SMS at no answer" [s]: (0 = off)

Day of deleting stats: (0 = off) Text of "SMS at no answer":

Minimal length of call after connect [s]: Text of SMS for all calls (number = %N):

Precision of counting length of call: CLIP to GSM separator:

(empty = off)

CLIP to GSM modification:

Use CLIP to GSM from INVITE field:

For proper functionality "Clip to GSM separator" has to be set

Save to the gate | Load from the gate | Default | Save to file

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.

File Gateway Gateway control Settings Help

Topics | Alphabetical glossary

- 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
- Restart

Groups assignment:

Module:	Outgoing:	Incoming:
0. module	<input type="text" value="1. Group"/>	<input type="text" value="1. Group"/>
1. module	<input type="text" value="1. Group"/>	<input type="text" value="1. Group"/>
2. module	<input type="text" value="2. Group"/>	<input type="text" value="1. Group"/>
3. module	<input type="text" value="2. Group"/>	<input type="text" value="1. Group"/>

Save to the gate | Load from the gate

Default | Save to file

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 configuration window for GSM incoming groups. The 'Mode' is set to 'Accept incoming calls + dialtone'. The 'List of called numbers' contains '101'. The 'Timeout for entering DTMF digit [s]' is set to 0. The 'Day of deleting GSM inc. group stastics' is set to 1. The 'CLIP to VoIP separator' is empty, with a red warning message: 'For proper functionality "Clip to VoIP separator" has to be set'. The 'Time to keep CLIP in table [hours]' is set to 0, and the 'Off' checkbox is checked. The 'Add record only for unconnected call' and 'Delete record for connected answer' checkboxes are unchecked.

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).

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 configuration interface for the Siemens HiPath 3000. The 'IP Trunks' tab is selected. The 'Selection' section shows 'Gatekeeper HG1500' and 'Slot 5'. The 'Trunks' section contains a table with the following data:

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

The right-hand panel shows the 'Number' dropdown menu set to 'IP Trunking', highlighted with a red box. Below it are 'Add', 'Delete', and 'Configured line' sections.

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 configuration window for a PBX route. The 'Routes' tab is active, and the 'interwork' route is selected. The 'Route Name' field contains 'interwork' and the 'Route prefix' field contains '859'. Below these are fields for 'CO code' (with a '2nd trunk code' checkbox), 'PABX number-incoming' (Country code, Local area code, PABX number), and 'PABX number-outgoing' (Country code, Local area code, PABX number). A 'Location number' checkbox is set to 'Trk Grp. 1'. The 'Digit transmission' dropdown menu is set to 'en-bloc sending' and is highlighted with a red box. The 'Numbering plan' section has radio buttons for 'Called Party Number' and 'All others', both with 'System check' selected. The 'Site' section has radio buttons for 'System check', 'Private network', and 'Always station', with 'System check' selected. The 'Switch' section contains several checkboxes: 'COLP' (checked), 'without CLIP', 'no DIV.LEG-Info', 'Always use DSP', 'Intern call like extern', 'Without CCNR', 'Notify send' (checked), and 'With sending complete' (unchecked). 'No SETUP ACK.' is also present. At the bottom are 'Reset', 'Apply', and 'Help' buttons.

2) LCR SETTING IN PBX

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

Example setting of Dialed digits:

OC6Z 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 V.8.0)

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.

■ 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 Node / IP Addresses

Node Number: 1

LAN trunking protocol: Native SIP

LAN Trunking type: Standard Trunking

HXG Gatekeeper Board 1 - IP Address: 192.168.1.50

HXG Board 2 - IP Address: 0.0.0.0

HXG Board 3 - IP Address: 0.0.0.0

HXG Board 4 - IP Address: 0.0.0.0

HXG Board 5 - IP Address: 0.0.0.0

HXG Board 6 - IP Address: 0.0.0.0

HXG Board 7 - IP Address: 0.0.0.0

HXG Board 8 - IP Address: 0.0.0.0

Alive Monitoring:

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)

PBX Node / IP Addresses

Node Number: 2

LAN trunking protocol: Native SIP

LAN Trunking type: Standard Trunking

HXG Gatekeeper Board 1 - IP Address: 192.168.1.120

HXG Board 2 - IP Address: 0.0.0.0

HXG Board 3 - IP Address: 0.0.0.0

HXG Board 4 - IP Address: 0.0.0.0

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 Lite
- 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
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- Destination Codec Parameters
- PBX
 - IP Networking Data
 - Nodes
 - 1
 - 2
 - Routing
 - 1 <1>
 - 6 <2>
 - 7 <2>
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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	Priority 2	VAD: <input type="checkbox"/>	30 msec
G.711 μ-law	Priority 7	VAD: <input type="checkbox"/>	30 msec
G.723	not used	VAD: <input type="checkbox"/>	30 msec
G.729A	Priority 1	VAD: <input type="checkbox"/>	20 msec
G.729AB	not used	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

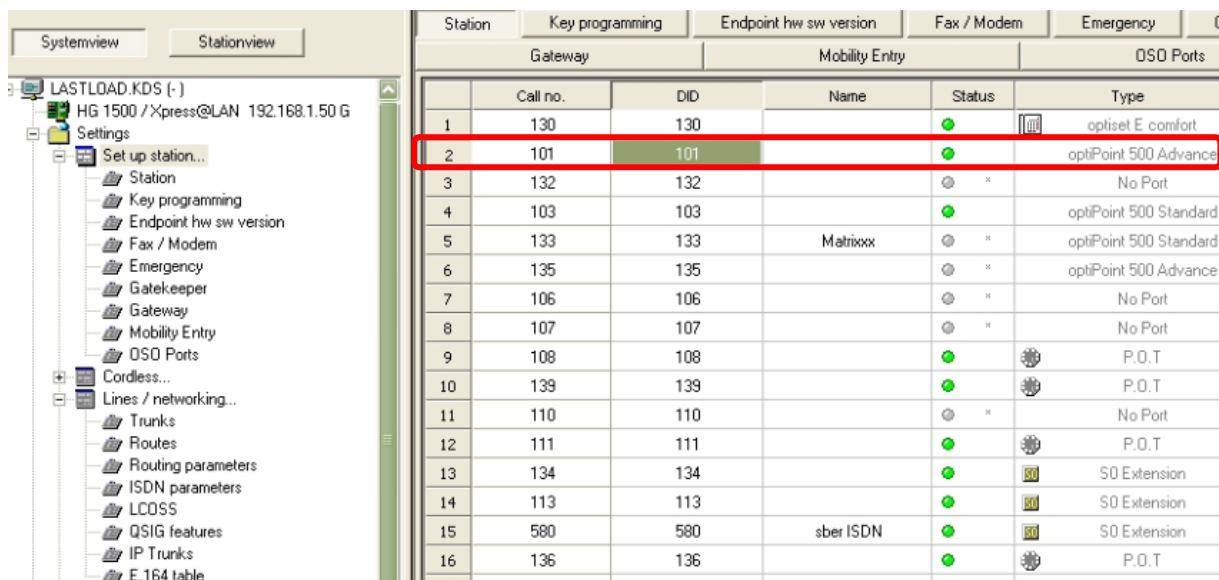
Misc.

ClearChannel: Frame Size: 20 msec

Incoming call from 2N® VoiceBlue Lite

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

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



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



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