



2N[®]

OfficeRoute



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

connected via SIP trunk

Quick guide

Version 1.00

www.2n.cz

2N® OfficeRoute 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

Settings of 2N® OfficeRoute

1) Switching SIP proxy off

Siemens HiPaths 3000 series can not change the destination port for SIP trunk. Default settings for destination port is 5060. 2N® OfficeRoute has port 5060 used by internal SIP proxy. That is why the SIP proxy has to be switched off for this case.



The screenshot displays the 2N OfficeRoute web interface. On the left is a navigation menu with the 2N logo and various settings categories. The main content area shows 'SIP proxy routing rules' with a table of rules. A red box highlights the 'Switch SIP Proxy off' option in the 'SIP proxy' section of the left menu.

If prefix	Strip	Add	Do action	With parameter			
sip:0	0		connect to LCR	SIP - Internal SIP line			<input type="checkbox"/>
sip:199	0		connect to LCR	SIP - Internal SIP line			<input type="checkbox"/>
else	0		lookup registration				

SIP proxy

- Switch SIP Proxy off
- Registrations

2) Creating a new SIP trunk

At menu **Telephony services** → **Devices** → **SIP lines** add a new line

The screenshot shows the 2N Telecommunications management interface. At the top, there is a navigation bar with icons for Network, User management, Telephony services (highlighted with a red box), Administration, States & Logs, and Messaging. On the left side, there is a sidebar menu with 'Devices' highlighted (red box) and sub-items: SIP lines (red box), H.323 lines, SIM cards, DISA lines, Fax lines, and Fxs line. Below these are other menu items: GSM, Services, Mobility Extension, Fax, LCR, GSM routing, and SIP proxy. The main content area is titled 'SIP lines' and contains a table with the following data:

Line ID	SIP server	Phone number	Listen port	Description			
[16]	192.168.1.1	vbegw	5061	Internal SIP line			<input type="checkbox"/>
[19]			5062				<input type="checkbox"/>

At the bottom right of the main content area, there is a red box around a plus sign icon and a checkmark icon, indicating the 'Add new' button.

For a new SIP line:

- Fill SIP server address as IP address of Siemens PBX, listen port 5060.
- Highlight or change a priority of codecs which should be used for this trunk.
- Uncheck Register with proxy checkbox.
- Write some description of this trunk.

Add SIP line

SIP server address:

SIP domain:

SIP name:

Display name:

Listen port:

If exists SIP proxy registration, listen port cannot be 5060

User name:

Password:

Codecs:

G.711 A Law 64000 bps

G.711 u Law 64000 bps

G.729 8000 bps

▲ Shift up

▼ Shift down

Add Phone context to REGISTER request:

Register expires (seconds):

Register with proxy:

Enable CLIP:

Allow only one call:

Get callee from:

Use Diversion header:

Enable NAT:

NAT port begin:

NAT port range:

NAT IP address:

No route code:

SIP TOS/DiffServ Value:

RTP TOS/DiffServ Value:

Description:

3) Configuration of the LCR (Least Cost Routing)

At **Telephony services** → **LCR** disable all rules which are not routed into GSM

LCR

Destination	Prefix	Enabled	Routes	Description
To FXS line 17	199	<input type="checkbox"/>	FXS line 17	factory default
To SIP line 16	1	<input type="checkbox"/>	SIP line 16	factory default
To DISA line 21	4	<input type="checkbox"/>	DISA line 21	factory default
To SIP line 19		<input type="checkbox"/>	SIP line 19	factory default
To GSM	.*	<input checked="" type="checkbox"/>	GSM	factory default

Remove all rules from Telephony services – LCR – Normalization

Normalization

Line	Prefix	Remove count	Add number	Type	Description
<input checked="" type="checkbox"/>	19	+	000	Caller incoming	factory default
<input type="checkbox"/>	19		0	Caller incoming	factory default
<input type="checkbox"/>	19		199	Called incoming	factory default
<input type="checkbox"/>		0		Called outgoing	factory default

4) Incoming calls

Incoming calls can be routed to some DDI in Siemens. For our case it is number 100 which will be dialled into our SIP trunk for all incoming calls. Create a new Operator rule in **Telephony services** → **GSM routing** → **Operator**

Operator number is a number which will be dialled into choosed VoIP line for incoming call.

Add GSM routing Operator service

Service name: Incoming calls

Operator number: 101

VoIP line: SIP - Siemens 3000 trunk

Description: Incoming calls from GSM

Assign the Operator service with GSM modules which should use this incoming rule. You can do it at **Telephony services** → **GSM routing** → **GSM**

Create here a new rule for each GSM module and choose the service

Modify GSM device routing "/dev/ttyS3"

GSM device: GSM module - 1

Service name: Incoming calls

Description: GSM1 to 101

Remove all normalization rules in GSM routing menu

GSM routing - Normalization

Prefix	Remove count	Add number	Description
<input checked="" type="checkbox"/>	+	000	factory default
<input type="checkbox"/>		0	factory default

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 displays the configuration interface for IP Trunks. 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' and 'Slot 5' selected, and a checked box for 'Enable gateway resources'. The main area is titled 'Trunks' and contains a table with 8 rows. The table has columns for Trunk, Code, Type, and Route. The 'Type' column shows 'SIP Provider 2' for lines 1-2 and 'IP Trunking' for lines 3-8. The 'Route' column shows 'SIP 2' for lines 1-2 and 'interwork' for lines 3-8. To the right of the table is a 'Number' section with a dropdown menu set to 'IP Trunking', an 'Add' button, a 'Selected line...' section with a 'Delete' button, and 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

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 system. The 'Routes' tab is active, and the 'interwork' route is selected. The 'Route Name' field contains 'interwork' and the 'Route prefix' field contains '859'. The 'CO code' section has a '2nd trunk code' checkbox. The 'PABX number-incoming' section includes fields for 'Country code', 'Local area code', and 'PABX number', along with a 'Location number' checkbox. The 'PABX number-outgoing' section has similar fields and a 'Suppress station number' checkbox. The 'Digit transmission' dropdown menu is set to 'en-bloc sending' and is highlighted with a red box. The 'Numbering plan' section has three sub-sections: 'Called Party Number', 'All others', and 'Site', each with radio button options for 'System check', 'ISDN numbering plan', 'Private numbering plan', and 'Unknown numbering plan'. The 'Switch' section at the bottom has 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'.

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® OfficeRoute 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® OfficeRoute gateway.

LAN trunking protocol needs to be “Native SIP” and IP address is the IP of 2N® OfficeRoute 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.

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® OfficeRoute
- 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

PBX Route Call Address

Node Number: 2

Station Number: 6

Service: Voice

Apply Undo

Codec setting

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

You can set up priorities for codecs.

The Siemens HiPath support DTMF via RFC 2833.

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): 1472

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

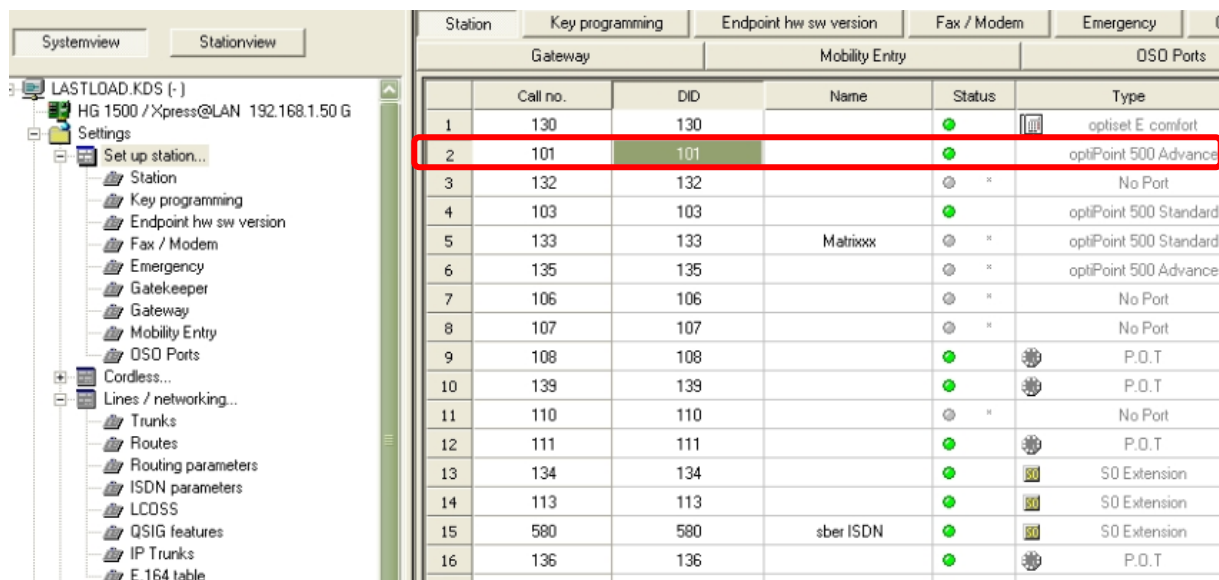
Misc.

ClearChannel: Frame Size: 20 msec

Incoming call from 2N® OfficeRoute

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

Now, the number coming from 2N® OfficeRoute 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



2N TELEKOMUNIKACE a.s.

Modřanská 621, 143 01 Praha 4
tel.: 261 301 111, fax: 261 301 999,
e-mail: sales@2n.cz
www.2n.cz