



2N®

VoiceBlue Next



2N® VoiceBlue Next & Cisco Unified Communications Manager Express verze 4.1(0)

connected via SIP trunk

Quick guide

Version 2.00

www.2n.cz

2N® VoiceBlue Next has these parameters:

- IP address 192.168.22.42
- Incoming port: 5060

Cisco Unified Communication Manager Express parameters:

- IP address 192.168.22.35
- 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 CME. 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 'Gateway' configuration page for 2N Telecommunications. The interface includes a sidebar with navigation options like 'Gateway control', 'Gateway configuration', and 'Configuration backup'. The main content area is titled 'Codec settings' and 'IP addresses'. Under 'IP addresses', there are fields for 'SIP proxy (IP->GSM)', 'SIP proxy (GSM->IP)', 'SIP registrar', 'NAT firewall', and 'STUN server'. The 'SIP proxy (GSM->IP)' field is highlighted with a speech bubble stating: 'The IP address and port which will accept traffic from'. The 'SIP proxy (IP->GSM)' field is also highlighted with a speech bubble stating: 'The IP address to which the traffic is send'. The 'SIP proxy (GSM->IP)' field is set to '192.168.22.35' and '5060'. The 'SIP proxy (IP->GSM)' field is set to '0.0.0.0' and '5060'. The 'SIP registrar' field is set to '0.0.0.0' and '5060'. The 'NAT firewall' field is set to '0.0.0.0'. The 'STUN server' field is set to '0.0.0.0' and '3478'. The 'Next STUN server request (60-6553, 0=off) [s]:' field is set to '6000'. The interface also includes a 'Logout' button and a 'Gateway' status bar with 'Update' and 'Restart' options.

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 left features the 2N TELECOMMUNICATIONS logo. The top right shows the 'Gateway' title with 'Update' and 'Restart' links. A left sidebar contains a 'Gateway control' section and a 'Gateway configuration' menu with options 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. The main content area is titled 'Prefixes' and includes a 'GSM prefix lists' section with tabs for Prefixlist 1 through 8. Below this is the 'Basic settings' section with fields for 'GSM network ID' and 'Default count of digits' (set to 9). The 'Table of replaced prefixes' and 'Table of accepted prefixes' sections both have a warning: 'Only 0123456789*#+ characters are allowed'. Each table has a list box, input fields for 'Prefix' and 'Replace with' (or 'Digits count'), and buttons for 'Add', 'Remove', and 'Remove all'. The bottom left has a 'Logout' button, and the bottom right has three document icons.

2N TELECOMMUNICATIONS

Gateway
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

Prefixes

GSM prefix lists

Prefixlist 1 | Prefixlist 2 | Prefixlist 3 | Prefixlist 4 | Prefixlist 5 | Prefixlist 6 | Prefixlist 7 | Prefixlist 8

Basic settings

GSM network ID:

Default count of digits:

Table of replaced prefixes

Only 0123456789*#+ characters are allowed

Prefix:

Replace with:

Add

Remove

Remove all

Table of accepted prefixes

Only 0123456789*#+ characters are allowed

Prefix:

[Digits count]:

Add

Remove

Remove all

Logout ⓘ

- 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 cards below to which GSM outgoing group.




Gateway

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 ⓘ

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:

Outgoing:

Incoming:

0. module	1. Group ▼	1. Group ▼
1. module	2. Group ▼	1. Group ▼

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)

2N TELECOMMUNICATIONS

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
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- Login configuration
- Web configuration
- Report configuration

Configuration backup

Logout

GSM outgoing groups

1. GSM group | 2. GSM group

General settings

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: ☒

Call length counting: Seconds

Disconnect call

SIM limit exceeded: ☐

Time limit exceeded: ☐

No ALERTING before CONNECT: ☐

Send CLIP from VoIP to GSM/UMTS

Attention! Must be supported by your GSM / UMTS operator. In other case outgoing calls to GSM / UMTS can be rejected!

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.

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.

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
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- Web configuration
- Report configuration

Configuration backup

1. GSM group

2. GSM group

GSM incoming groups

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)

Logout

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

List of called numbers

Only 0123456789*#+ characters are allowed

100

Add

Remove

Remove all

CISCO UNIFIED COMMUNICATION MANAGER EXPRESS SETTING

For configuration was used freeware program - Cisco Configuration Professional - version 2.3.

For configuration, enter the menu: **Configure → Voice → Dial Plan → VoIP** and set up the prefix and IP address to the 2N VoiceBlue Next gateway.

The prefix **6.T** as in the example means that prefix is 6 plus other digits after 6 without limit.

The screenshot shows the 'Digit Manipulation' tab in the Cisco Configuration Professional interface. The configuration is for a dial peer with the following settings:

- Dial Peer Number *:** 2
- Description :** 2N_VoiceBlue_Next (1-64 Characters)
- Priority :** Priority 0
- Remote Site :** ☒ 192.168.22.42 ☐ SIP Trunk
- Destination Number :** 6.T
- Incoming Called Number :** (empty field)
- Answer Address :** (empty field)
- Shutdown Dial Peer :** ☒ No ☐ Yes
- Protocol :** ☐ H.323 ☒ SIP
- Codec :** ☒ g711alaw ☐ Voice Class Codec 1
- DTMF Tone Relay Type :** rtp-nte

Voice Activity Detection

With voice activity detection enabled, only speech voice data packets are sent over the network, and silence voice data packets are dropped optimizing the network bandwidth usage.

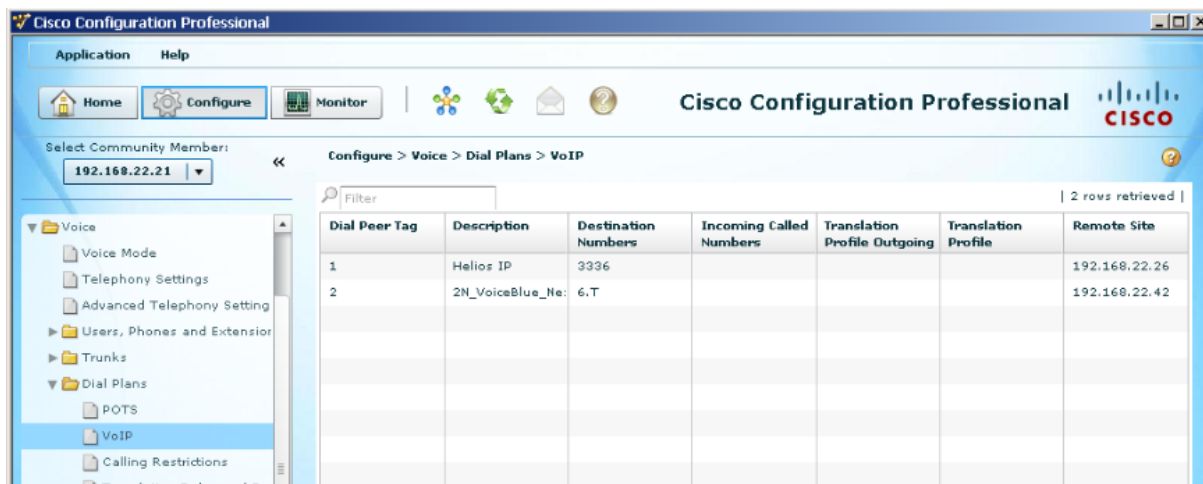
☒ Enable voice activity detection

* Indicates a mandatory field

OK

Cancel

In the picture below, you can see the configuration program with the saved routing to the 2N VoiceBlue Next.



Incoming calls are automatically enabled by a new trunk. All incoming calls to from 2N VoiceBlue Next will be routed to stations in CME or you can create your own dial plan.

In the CME version 4.1 you are not able to register SIP phones.



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