

CCM Express version 4.1 - How to interconnect with CCM Express version 4.1?

- 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

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 for secure communication with traffic from your CME only. 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 configuration' page for a 2N Gateway. The 'IP addresses' section is highlighted, showing the following settings:

Service	IP Address	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
STUN server:	0.0.0.0	3478	Set default port

Two callouts provide additional context:

- Callout 1:** "The IP address to which the traffic is send" points to the IP address field in the SIP proxy (IP→GSM) row.
- Callout 2:** "The IP address and port from which SIP packets will be accepted" points to the IP address and port fields in the SIP proxy (GSM→IP) row.

2. Configuration of the LCR (Least Cost Routing)

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

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 ⓘ

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
/	

Prefix:

Replace with:

Add

Remove

Remove all

Table of accepted prefixes

Only 0123456789*#+ characters are allowed

Prefix
6
7

Prefix:

[Digits count]:

Add

Remove

Remove all

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.



2N TELECOMMUNICATIONS Gateway | Update | Restart

Gateway control

Gateway configuration

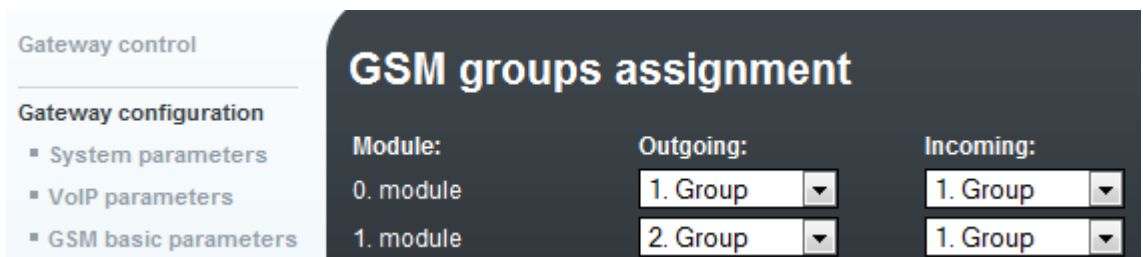
- System parameters
- VoIP parameters
- GSM basic parameters
- GSM groups assignment
- GSM outgoing groups
- GSM incoming groups
- Prefixes
- LCR table**
- CLIP Routing table
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LCR table

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 ▼

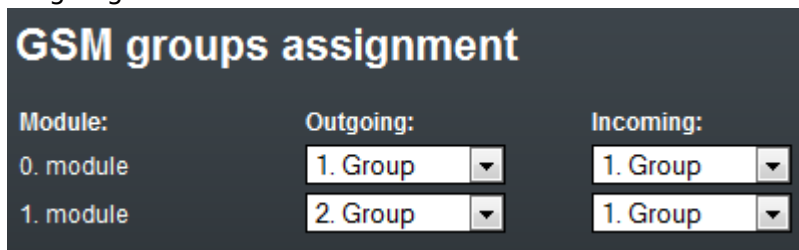
- Configuration of GSM outgoing groups:
You are able to set up different settings for each GSM group (CLIR, free minutes, virtual ring tone, roaming and others).



The screenshot shows the 'Gateway' configuration page for 'GSM outgoing groups'. The interface includes a sidebar with navigation options like 'Gateway control', 'Gateway configuration', and 'Configuration backup'. 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 options with dropdown menus: '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' (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' option and a red warning message: 'Attention! Must be supported by your GSM / UMTS operator. In other case outgoing calls to GSM / UMTS can be rejected!'. The bottom right corner has icons for save, refresh, and print.

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.



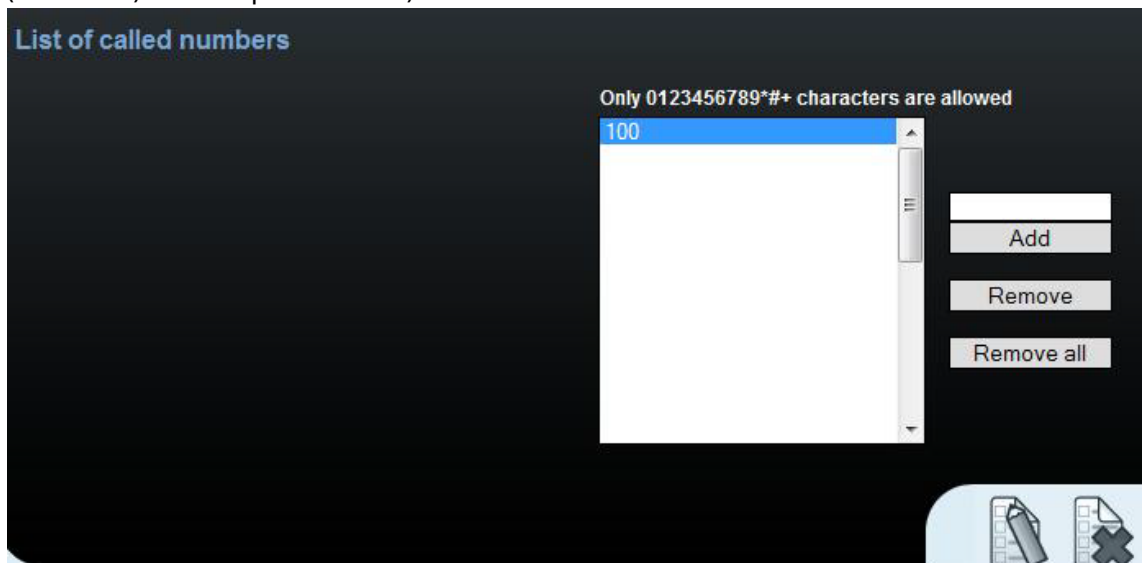
The screenshot shows the 'GSM groups assignment' configuration table. It has three columns: 'Module:', 'Outgoing:', and 'Incoming:'. There are two rows of data.

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.

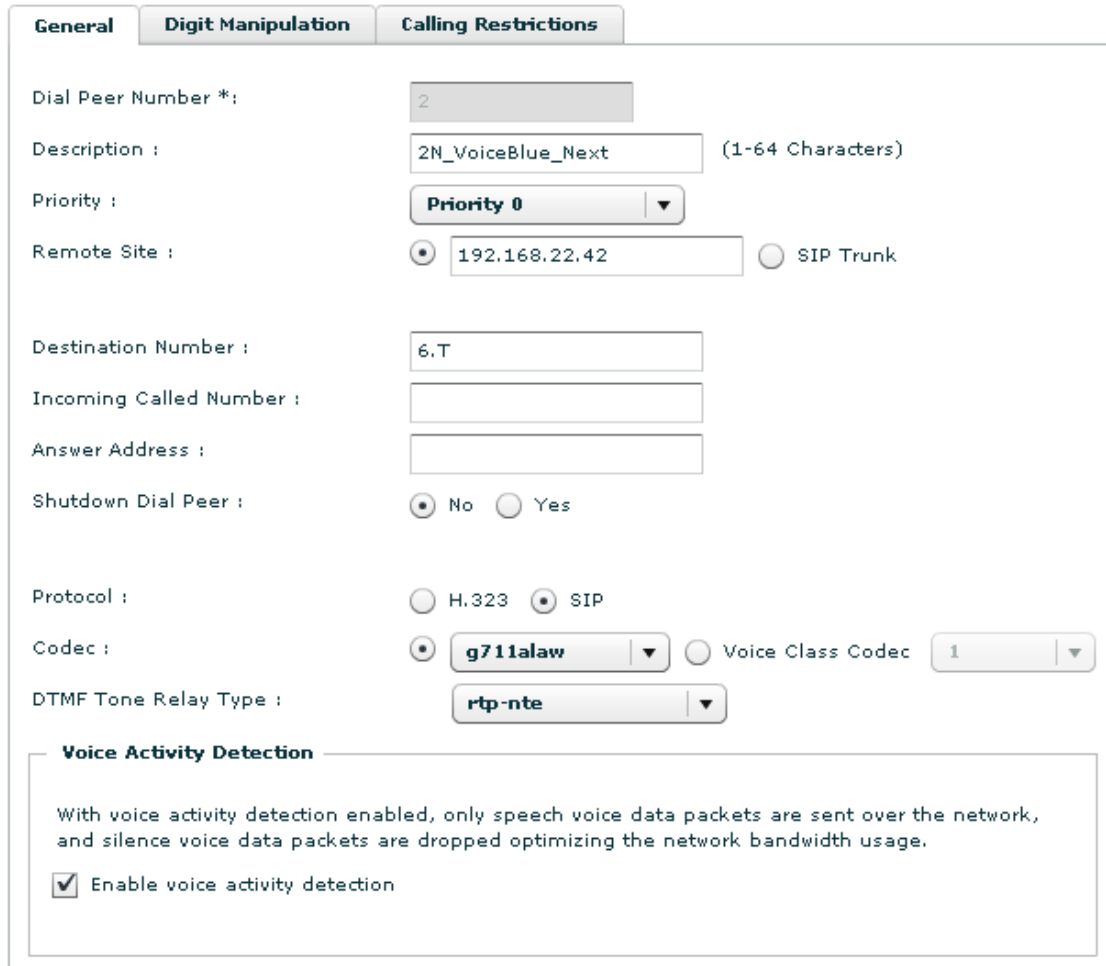


You can define the list of called numbers which will be automatically dialed after DTMF dialing timeout if the customer does not press any button within the specified time. From the configuration you can see 10 seconds for DTMF dialing and after that the call will be routed to the extension 100 to your CME (if you set up SIP proxy (GSM->IP) in VoIP parameters).



Cisco Unified Communication Manager Express setting

- Freeware program was used for the configuration- 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 in the example means that prefix is 6 plus other digits after 6 without limit.



The screenshot shows the 'Dial Peer' configuration window in Cisco Configuration Professional. The 'Calling Restrictions' tab is active. The configuration includes:

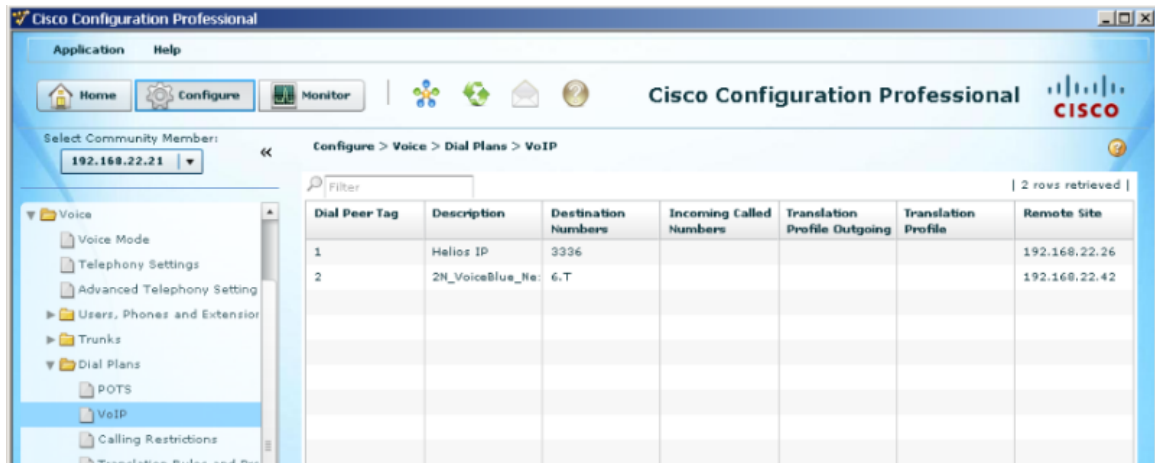
- Dial Peer Number *:** 2
- Description :** 2N_VoiceBlue_Next (1-64 Characters)
- Priority :** Priority 0
- Remote Site :** 192.168.22.42 SIP Trunk
- Destination Number :** 6.T
- Incoming Called Number :** (empty)
- Answer Address :** (empty)
- Shutdown Dial Peer :** No Yes
- Protocol :** H.323 SIP
-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

* 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 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.

!!! Download the PDF by clicking on the arrow on the left side!!!

More product information:

2N[®] VoiceBlue Next ([Official Website 2N](http://www.2n.cz))