



2N<sup>®</sup>

# VoiceBlue Next



## **2N<sup>®</sup> VoiceBlue Next & CISCO (CUCM 6, 7, 8)**

connected via SIP trunk

Quick guide

Version 4.00

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

## 2N® VoiceBlue Next has these parameters:

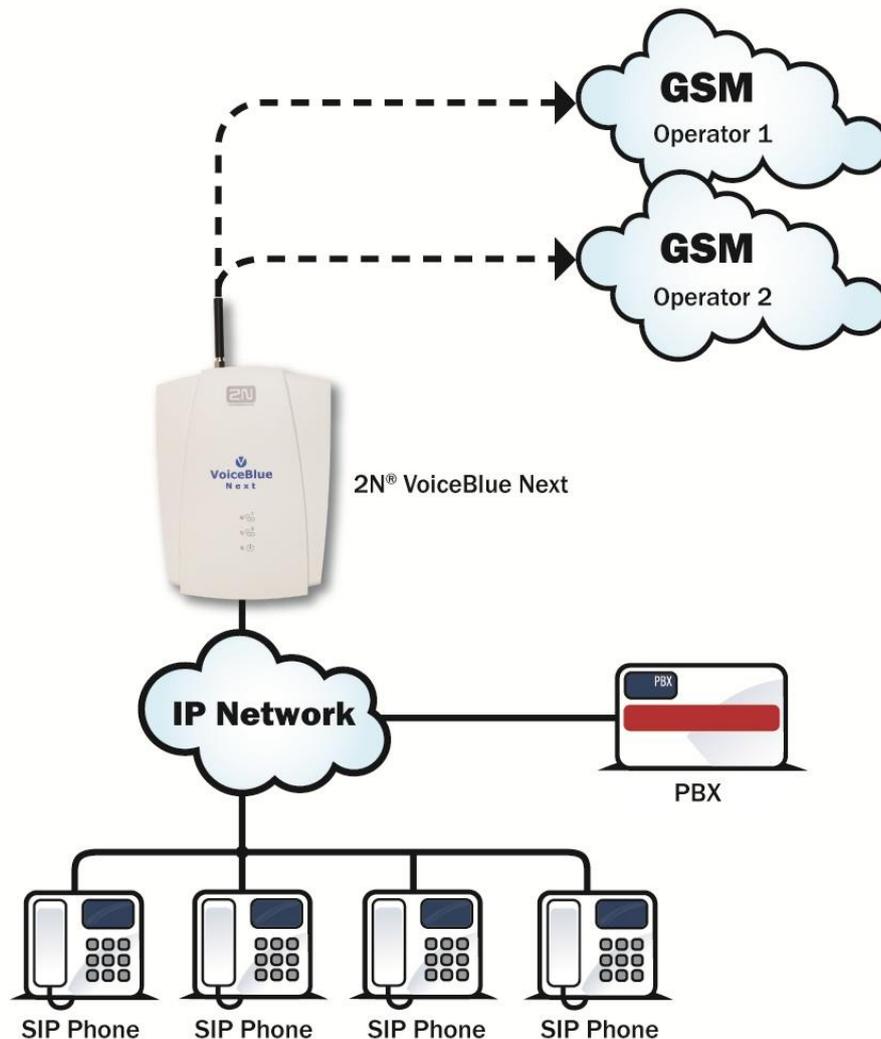
- IP address 192.168.22.42
- Incoming port: 5060
- Firmware: 01.00.04

## Cisco CallManager parameters:

- IP address 192.168.22.35
- Incoming port: 5063
- Firmware: CUCM 8.0

## Scenario

If we have an IP network in which an CiscoCall Manager, several SIP phones and 2N® VoiceBlue Next are connected, the configuration would be as shown in the figure below. Furthermore, suppose that the network is addressed as shown in the figure and GSM numbers are all numbers starting with 6, 7 and containing 9 digits.



## SIP TRUNK INTERCONNECTION

- 1) For the setting of the trunk between the VoiceBlue Next and your CUCM, you need to configure "SIP proxy (GSM→IP)" for GSM incoming calls. "SIP proxy (IP→GSM)" is designed only for secure communication with the traffic from your CUCM. You can specify the IP address and port where the IP packets will be accepted.

The screenshot shows the 'Gateway' configuration page for '2N TELECOMMUNICATIONS'. The page has a navigation bar with 'Gateway | Update | Restart' and a sidebar menu. The main content area is titled 'Gateway control' and includes sections for 'Codec priority' and 'IP addresses'. Two callout boxes provide context: one points to the 'SIP proxy (IP->GSM)' field with the text 'The IP address where the traffic is sent', and another points to the 'SIP proxy (GSM->IP)' field with the text 'The IP address and port which the traffic will come from'.

Field	Value	Port	Action
SIP proxy (IP->GSM):	0.0.0.0	5063	Set default port
SIP proxy (GSM->IP):	192.168.22.35	5063	Set default port
SIP registrar:	0.0.0.0	5060	Set default port
NAT firewall:	0.0.0.0		
STUN server:	0.0.0.0	3478	Set default port
Next STUN server request (60-6553, 0=off) [s]:	6000		

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

You have to specify prefixes for the operators in the country you are currently located. An example of this would be that in Czech Republic prefix 6 and 7 have a 9 digits number.

The setting is displayed below.

3) You need to create specific guidelines connecting prefixes with the GSM group. In the “GSM group” you will specify settings for SIM cards assigned to this specific group. In the “GSM group assignment” you can assign the module for the appropriate GSM outgoing group.

The screenshot shows the 2N Gateway control interface. On the left is a navigation menu with 'Gateway control' and 'Gateway configuration' sections. The 'Gateway configuration' section includes 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 'LCR table' option is selected, displaying a table with the following data:

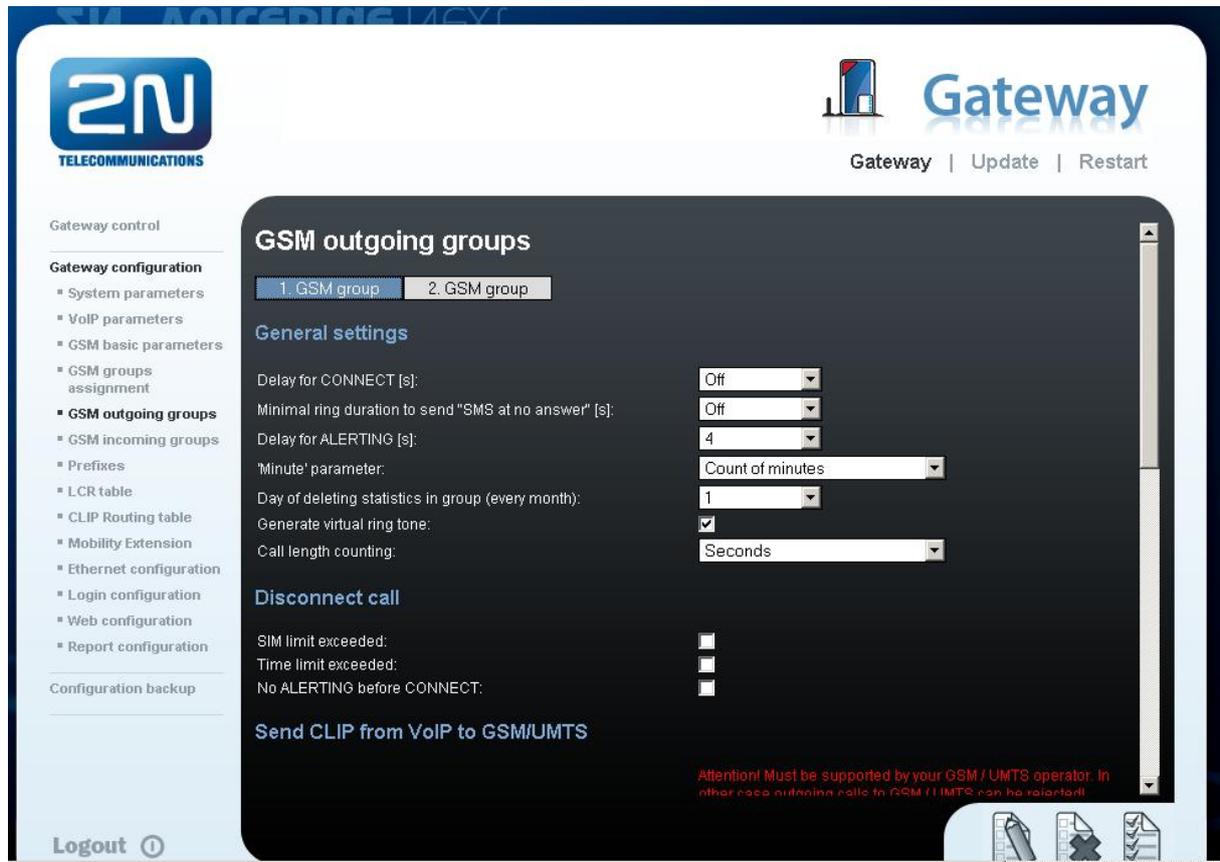
Prefix list	Time limitation	Weekend usage	Max. length of call	Groups	Add	Remove all
1/	0:00/24:00	Use as in week	Off	1	Edit	Remove
2/	0:00/24:00	Use as in week	Off	2	Edit	Remove

At the bottom left of the interface is a 'Logout' button with an information icon. At the bottom right are icons for a clipboard, a document with an 'X', and a document with a checkmark.

The screenshot shows the 'GSM groups assignment' configuration screen. It features a 'Module:' section with two entries: '0. module' and '1. module'. To the right is an 'Outgoing:' section with two dropdown menus. The first dropdown is set to '1. Group' and the second is set to '2. Group'. To the right of these is an 'Incoming:' section with two dropdown menus, both set to '1. Group'. A red circle highlights the 'Outgoing:' dropdowns.

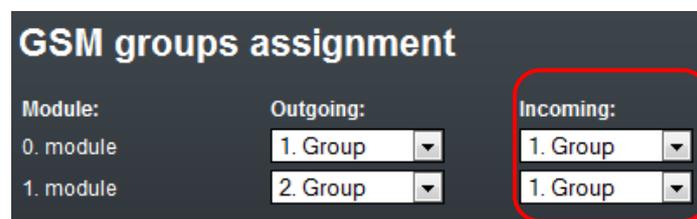
#### 4) Configuration of GSM outgoing groups:

You are able to have different setting for each GSM group (CLIR, free minutes, Virtual ring tone, roaming and others)



#### 5) Incoming calls

For incoming calls you can define 2 groups with the different behaviors and assign them to the GSM modules. The settings are similar with "GSM groups assignment" for outgoing calls.



In GSM incoming groups you can specify the traits for each GSM incoming group. Choose the mode to Reject, Ignore, Accept incoming calls or Callback.

The screenshot shows the 'Gateway' configuration interface. At the top left is the '2N TELECOMMUNICATIONS' logo. At the top right is the 'Gateway' logo with 'Update' and 'Restart' links. The main content area is titled 'GSM incoming groups' and has two tabs: '1. GSM group' (selected) and '2. GSM group'. Under 'General settings', there are several configuration options:
 

- Mode: Accept incoming calls + dialtone (dropdown)
- (Call number by %A, %G95..8 or none or answer and wait for DTMF)
- Minimum digits in DTMF: 4 (dropdown)
- Maximum digits in DTMF: 9 (dropdown)
- DTMF dialling timeout [s]: 10 (dropdown)
- Day of deleting GSM inc. group statistics (every month): 1 (dropdown)
- Prefix before DISA dial-in: [text input]
- CLIP (-' removes one digit): [text input]
- Looping of voice message [min]: Off (dropdown)

 Below this is the 'Send CLIP from GSM/UMTS to VoIP' section with:
 

- Transfer CLIP from GSM/UMTS: [checkbox]
- Separating char: [text input]
- Modify (-' removes one digit): [text input] (All groups)

 A sidebar on the left contains a navigation menu with categories like 'Gateway control', 'Gateway configuration', and 'Configuration backup'. At the bottom left is a 'Logout' button. At the bottom right are icons for editing and deleting.

You can define the list of numbers called. The number will be automatically dialed after the DTMF dialing has timed out. This happens when the customer doesn't press any button until the specific time. At this point, the number will be routed to the extension 100 to your CiscoCall Manager (if you set up SIP proxy (GSM->IP) in VoIP parameters).

The screenshot shows the 'List of called numbers' configuration interface. At the top, it states 'Only 0123456789\*#+ characters are allowed'. Below this is a list box containing the number '100'. To the right of the list box are three buttons: 'Add', 'Remove', and 'Remove all'. At the bottom right are icons for editing and deleting.

# CISCO CALL MANAGER SETTING

## 1) Create a new trunk

Add new trunk in the menu Device → Trunk → Add new

You need to set up Trunk Type: **SIP Trunk** and Device Protocol: **SIP**

The screenshot shows the Cisco Unified CM Administration interface. The navigation menu includes System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Trunk Configuration' and includes a 'Next' button. The 'Status' section shows 'Status: Ready'. The 'Trunk Information' section is highlighted with a red box and contains the following fields:

Trunk Type*	SIP Trunk
Device Protocol*	SIP

Below the form, there is a 'Next' button and a note: '\* - indicates required item.'

## Set up the trunk as in the picture below

The screenshot shows the Cisco Unified CM Administration interface for the 'Device Information' page. The navigation menu is the same as in the previous screenshot. The main content area is titled 'Trunk Configuration' and includes buttons for Save, Delete, Reset, Apply Config, and Add New. The 'Status' section shows 'Status: Ready'. The 'Device Information' section contains the following fields:

Product:	SIP Trunk
Device Protocol:	SIP
Device Name*	2N_VoiceBlue_Next
Description	2N_VoiceBlue_Next
Device Pool*	Testing
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_TestTeam
Location*	Hub_None
AAR Group	< None >
Packet Capture Mode*	None
Packet Capture Duration	0

Below the form, there are several checkboxes and a dropdown menu:

- Media Termination Point Required
- Retry Video Call as Audio
- Transmit UTF-8 for Calling Party Name
- Unattended Port
- SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.
- Use Trusted Relay Point\* Default

**Calling Search Space** is setting for incoming call to CCM. In the TestTeam is set up what numbers are allowed to call.

**AAR (Alternative Call Routing)** is not necessary to set up. It depends on the customer requirement.

**Incoming Calling Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Number Type	Prefix	Strip Digits	Use Device Pool CSS	Ca
Unknown Number	Default	0	<input checked="" type="checkbox"/>	< None >

**Multilevel Precedence and Preemption (MLPP) Information**

MLPP Domain: < None >

**Call Routing Information**

Remote-Party-Id  
 Asserted-Identity  
Asserted-Type\*: Default  
SIP Privacy\*: Default

**Inbound Calls**

Significant Digits\*: All  
Connected Line ID Presentation\*: Default  
Connected Name Presentation\*: Default  
Calling Search Space: **TestTeam**  
AAR Calling Search Space: TestTeam  
Prefix DN:   
 Redirecting Diversion Header Delivery - Inbound

**Destination address** is the IP address of the 2N® VoiceBlue Next.

**SIP Trunk Security Profile** have to be set as: Non Secure SIP Trunk Profile and Outgoing Transport type must be set up for UDP communication

**Outbound Calls**

Called Party Transformation CSS: TestTeam  
 Use Device Pool Called Party Transformation CSS  
Calling Party Transformation CSS: TestTeam  
 Use Device Pool Calling Party Transformation CSS  
Calling Party Selection\*: Originator  
Calling Line ID Presentation\*: Default  
Calling Name Presentation\*: Default  
Caller ID DN:   
Caller Name:   
 Redirecting Diversion Header Delivery - Outbound

**SIP Information**

Destination Address: **192.168.22.26**  
Destination Address IPv6:   
 Destination Address is an SRV  
Destination Port\*: 5060  
MTP Preferred Originating Codec\*: 711ulaw  
Presence Group\*: Standard Presence group  
SIP Trunk Security Profile\*: **Non Secure SIP Trunk Profile**  
Rerouting Calling Search Space: TestTeam  
Out-Of-Dialog Refer Calling Search Space: TestTeam  
SUBSCRIBE Calling Search Space: TestTeam  
SIP Profile\*: Standard SIP Profile  
DTMF Signaling Method\*: RFC 2833

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | Go

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**SIP Trunk Security Profile Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Copy | Reset | Apply Config | Add New

**Status**  
Status: Ready

**SIP Trunk Security Profile Information**

Name\*: 2N Non Secure SIP Trunk Profile  
 Description: Non Secure SIP Trunk Profile authenticated by null S  
 Device Security Mode: Non Secure  
 Incoming Transport Type\*: TCP+UDP  
 Outgoing Transport Type: **UDP**  
 Enable Digest Authentication  
 Nonce Validity Time (mins)\*: 600  
 X.509 Subject Name:  
 Incoming Port\*: 5060  
 Enable Application Level Authorization  
 Accept Presence Subscription  
 Accept Out-of-Dialog REFER\*\*  
 Accept Unsolicited Notification  
 Accept Replaces Header  
 Transmit Security Status

Save | Delete | Copy | Reset | Apply Config | Add New

## 2) Route pattern settings

Enter the menu **Call routing** → **Route/Hunt** → **Route Pattern** → **Select gateway**

You can use route list and groups but just for one gateway connection you don't need to have it. You can simply set up rules in Route pattern.

### **Examples of Route pattern:**

724! means prefix 724 and all other digits after

724XXXXXX means prefix 724 and 6 more digits

0.6! means that dialled 0 will be striped and 6 with other digits can be dialled. In the Called Party Transformation must be set up PreDot for the stripping of the 0.

**Route Pattern Configuration** Related Links: [Back](#)

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**Status**

Status: Ready

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**Pattern Definition**

Route Pattern\*   
Route Partition   
Description   
Numbering Plan   
Route Filter   
MLPP Precedence\*   
Resource Priority Namespace Network Domain   
Route Class\*   
Gateway/Route List\*  [\(Edit\)](#)  
Route Option  
 Route this pattern  
 Block this pattern   
Call Classification\*   
 Allow Device Override
 Provide Outside Dial Tone
 Allow Overlap Sending
 Urgent Priority  
 Require Forced Authorization Code  
Authorization Level\*   
 Require Client Matter Code

- **If you make any any change, you need to save it and then apply changes!!**

**Calling Party Transformations**

Use Calling Party's External Phone Number Mask  
Calling Party Transform Mask   
Prefix Digits (Outgoing Calls)   
Calling Line ID Presentation\*   
Calling Name Presentation\*   
Calling Party Number Type\*   
Calling Party Numbering Plan\*

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**Connected Party Transformations**

Connected Line ID Presentation\*   
Connected Name Presentation\*

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**Called Party Transformations**

Discard Digits   
Called Party Transform Mask   
Prefix Digits (Outgoing Calls)   
Called Party Number Type\*   
Called Party Numbering Plan\*

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**ISDN Network-Specific Facilities Information Element**

Network Service Protocol   
Carrier Identification Code   
Network Service 
Service Parameter Name 
Service Parameter Value



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