2N®

VoiceBlue Next

2N® VoiceBlue Next & CISCO (CUCM 6, 7, 8)

connected via SIP trunk

Quick guide

Version 4.00 **www.2n.cz**

2N® VoiceBlue Next has these parameters:

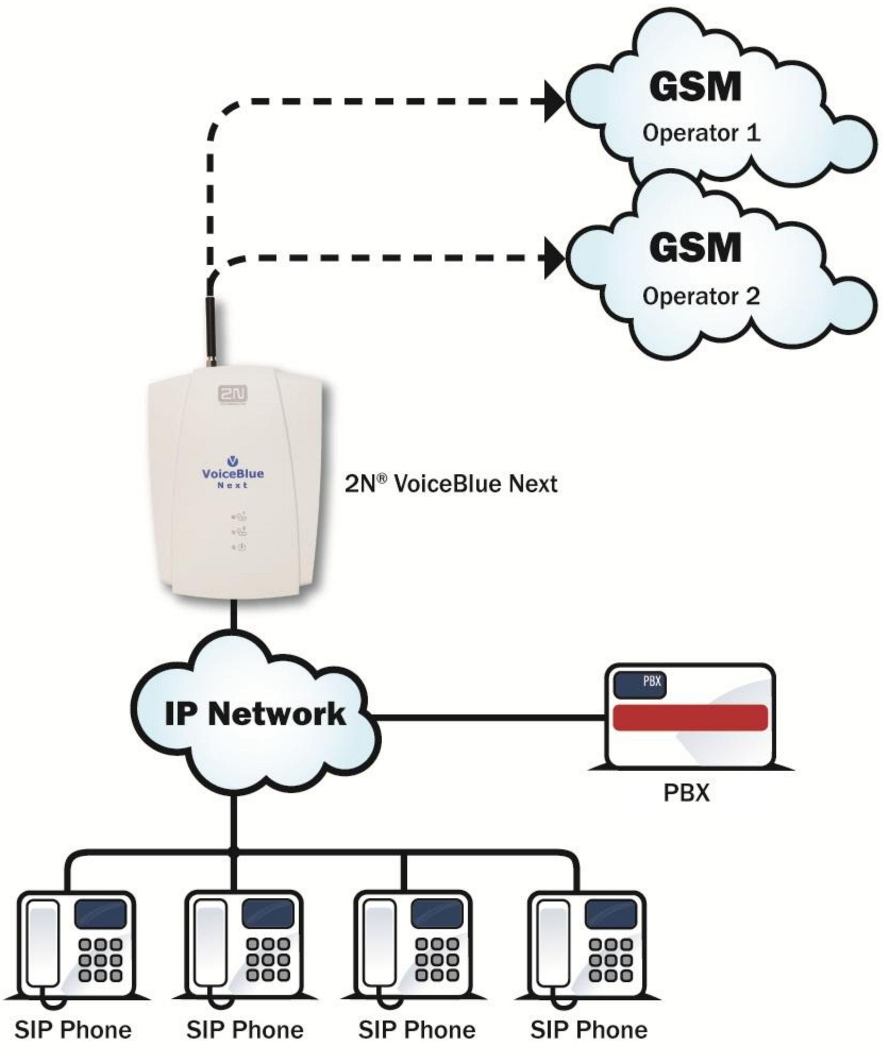
- IP address 192.168.22.42
- Incoming port: 5060
- Firmware: 01.00.04

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

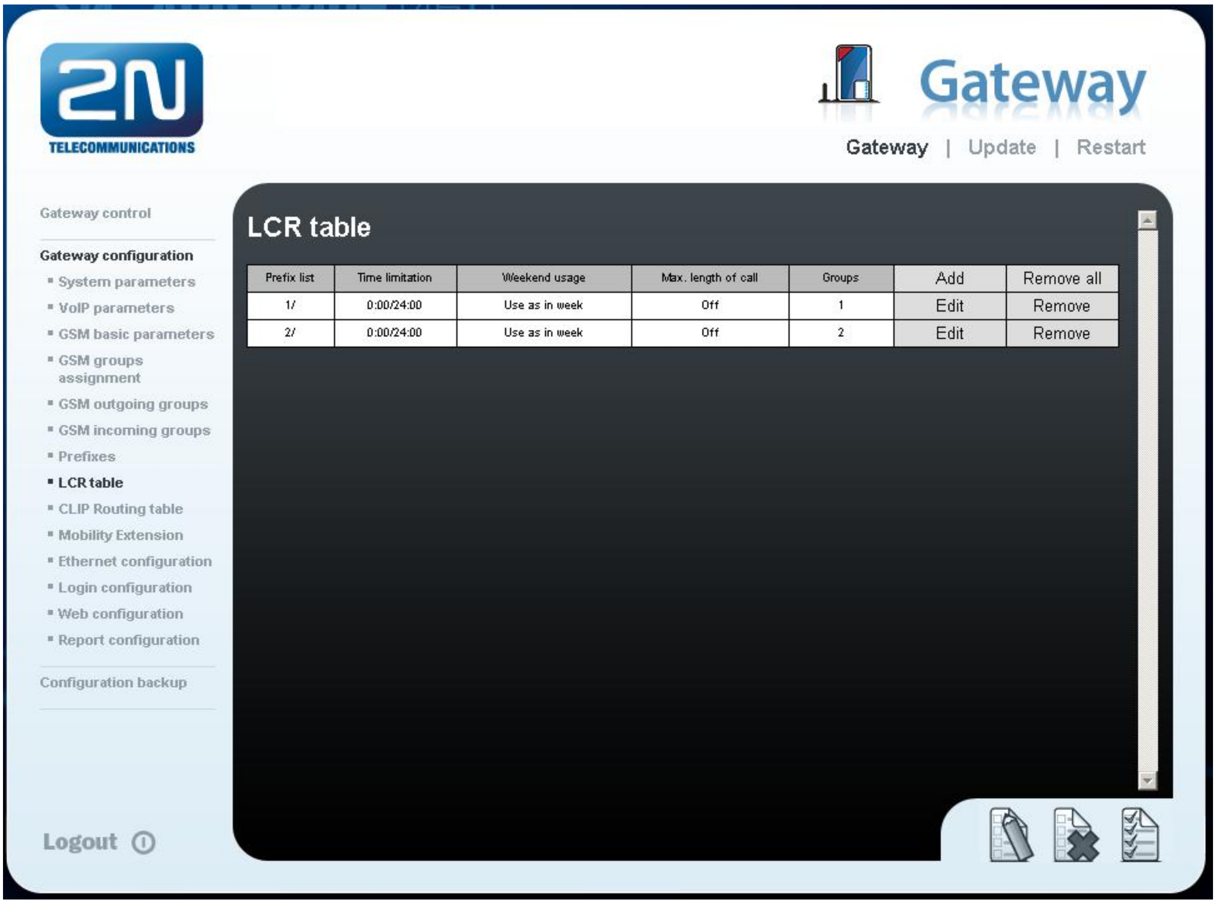
1) For the setting of the trunk between the VoiceBlue Next and your CUCM, you need to configure “SIP proxy (GSM→IP)” for GSM incoming calls. “SIP proxy (IP→GSM)” is designed only for secure communication with the traffic from your CUCM. You can specify the IP address and port where the IP packets will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

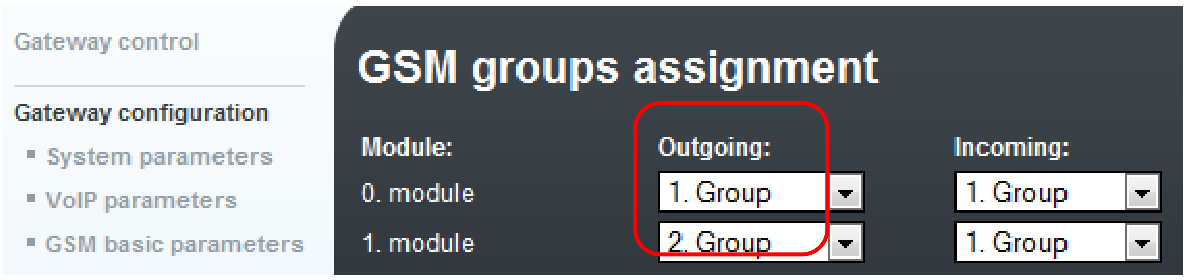
2) Configuration of the LCR (Least Cost Routing)

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.

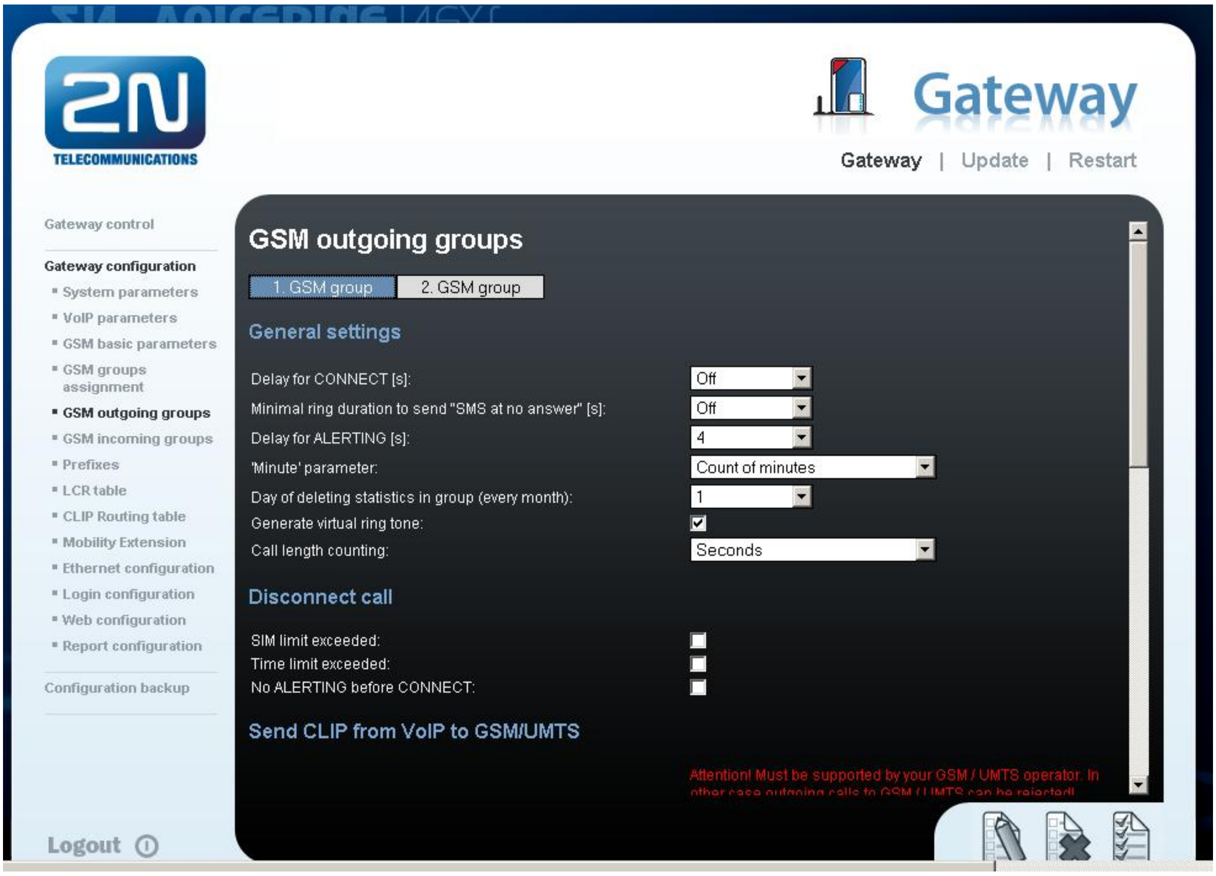
3) You need to create specific guidelines connecting prefixes with the GSM group. In the "GSM group" you will specify settings for SIM cards assigned to this specific group. In the "GSM group assignment" you can assign the module for the appropriate GSM outgoing group.





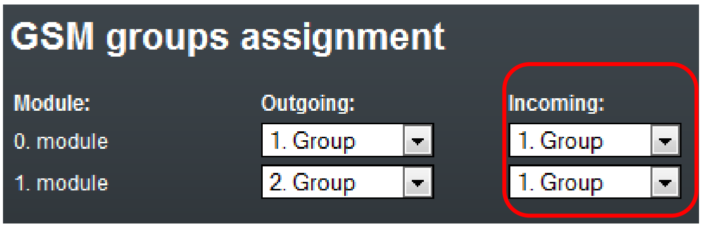
4) Configuration of GSM outgoing groups:

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



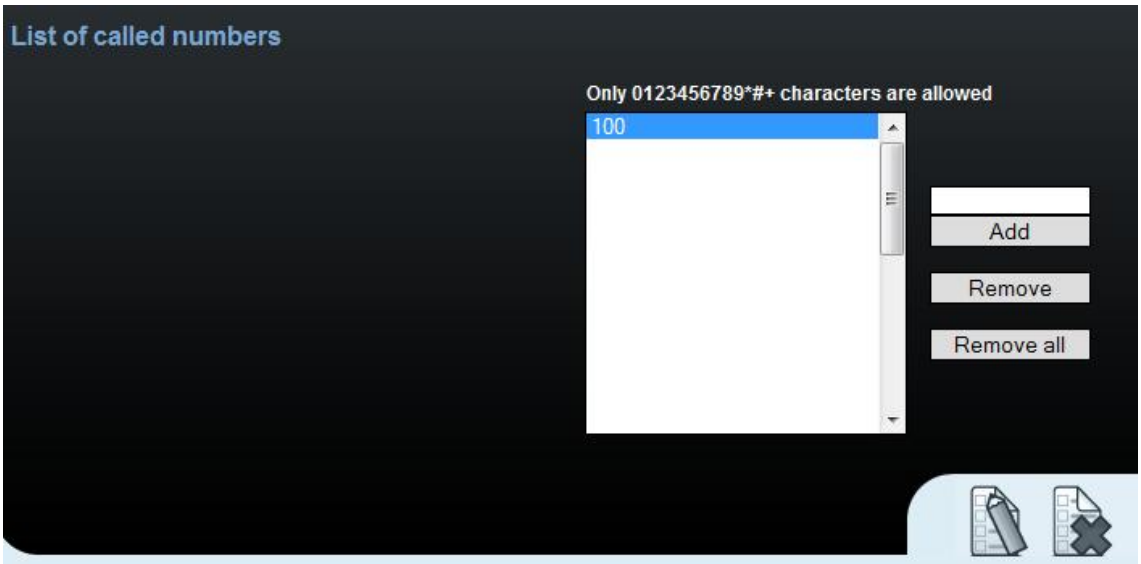
5) Incoming calls

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.

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).

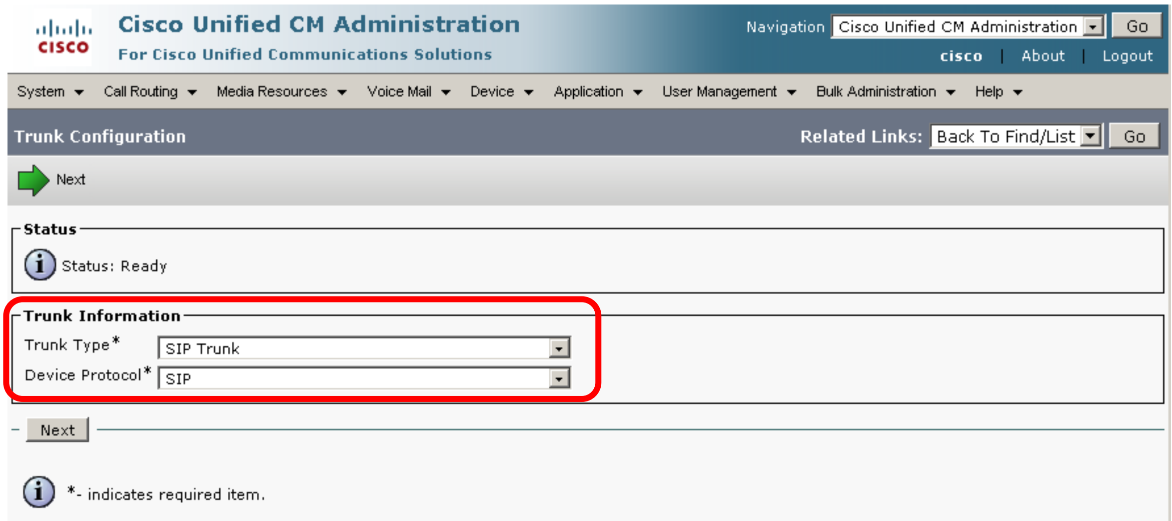


CISCO CALL MANAGER SETTING

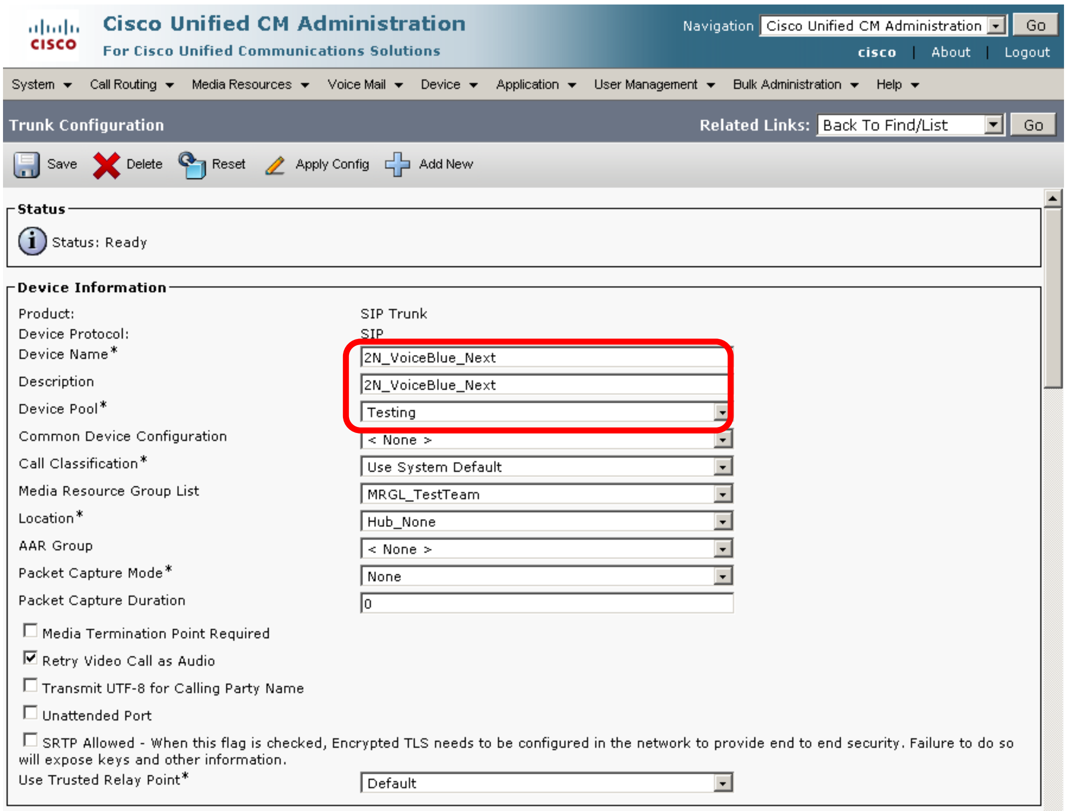
1) Create a new trunk

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

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

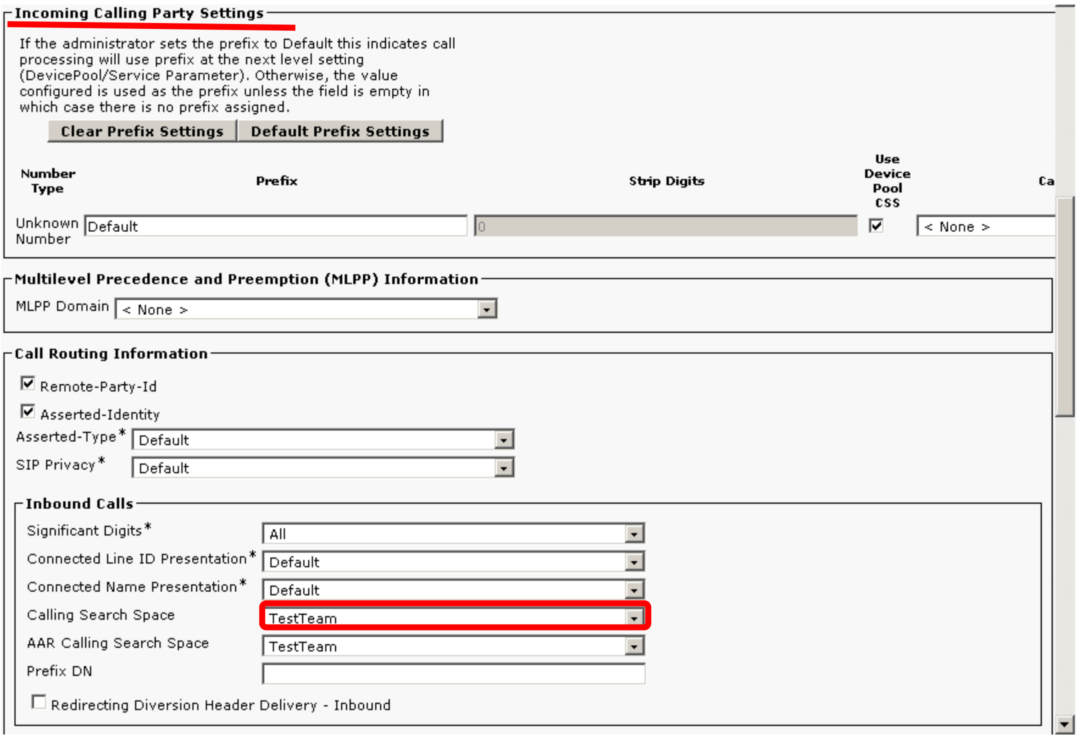


Set up the trunk as in the picture below



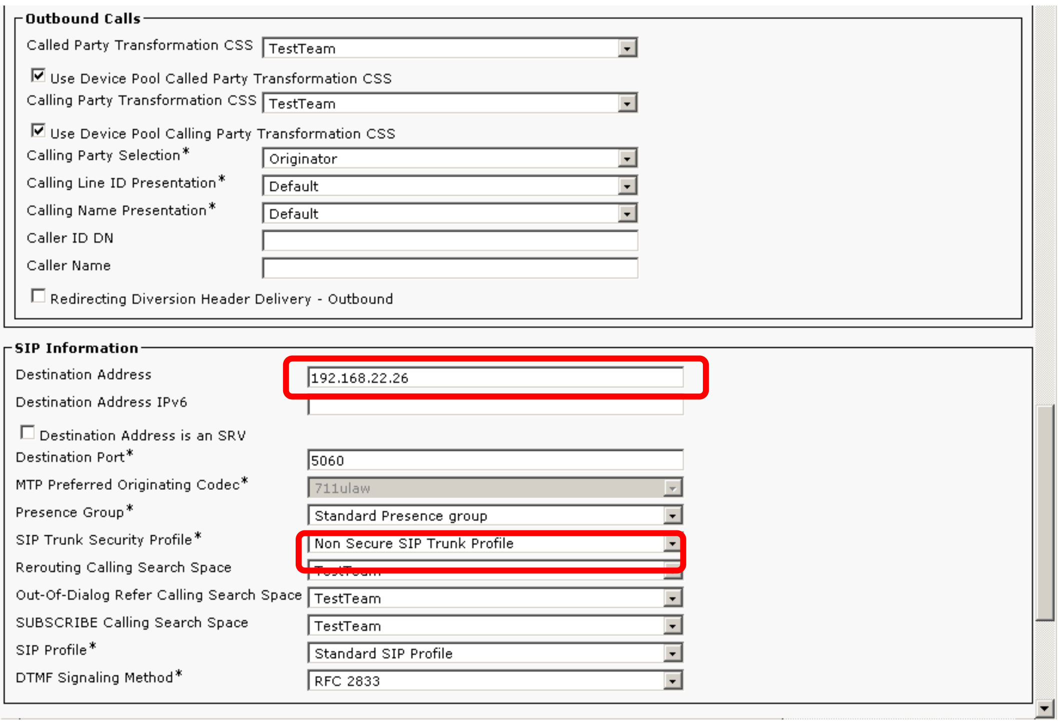
Calling Search Space is setting for incoming call to CCM. In the TestTeam is set up what numbers are allowed to call.

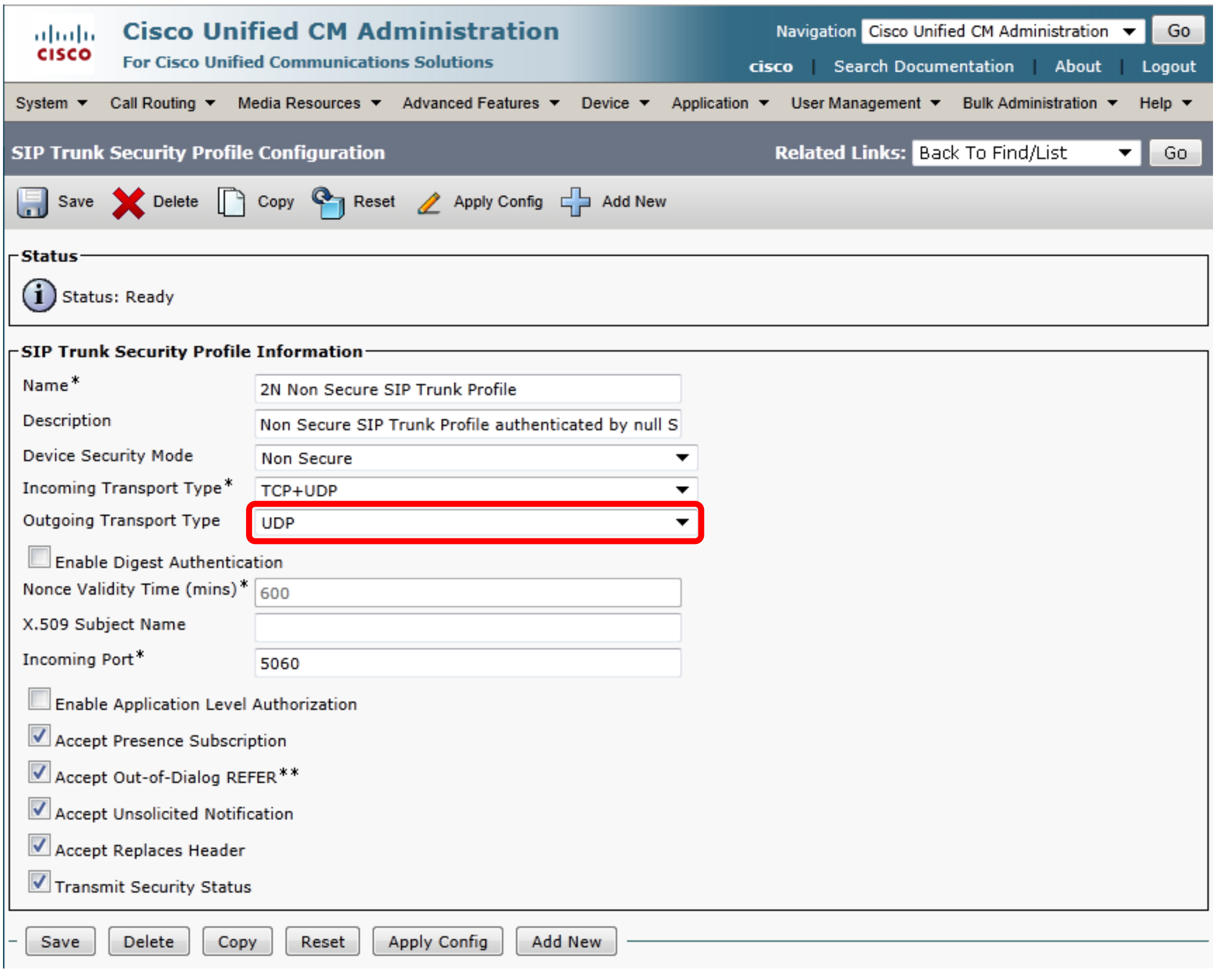
AAR (Alternative Call Routing) is not neccesary to set up. It depends on the customer requirement.



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





2) Route pattern settings

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

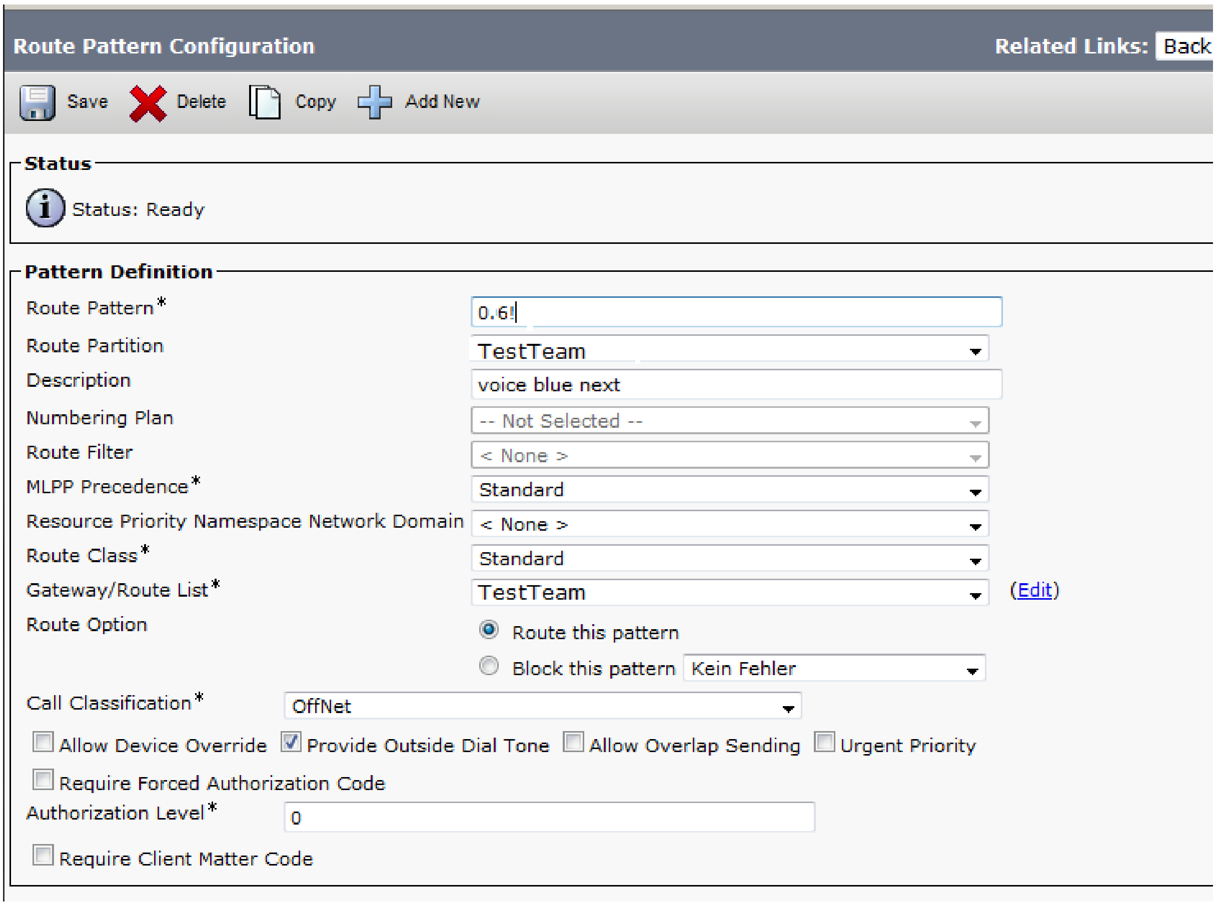
You can use route list and groups but just for one gateway connection you don´t need to have it. You can simply set up rules in Route pattern.

Examples of Route pattern:

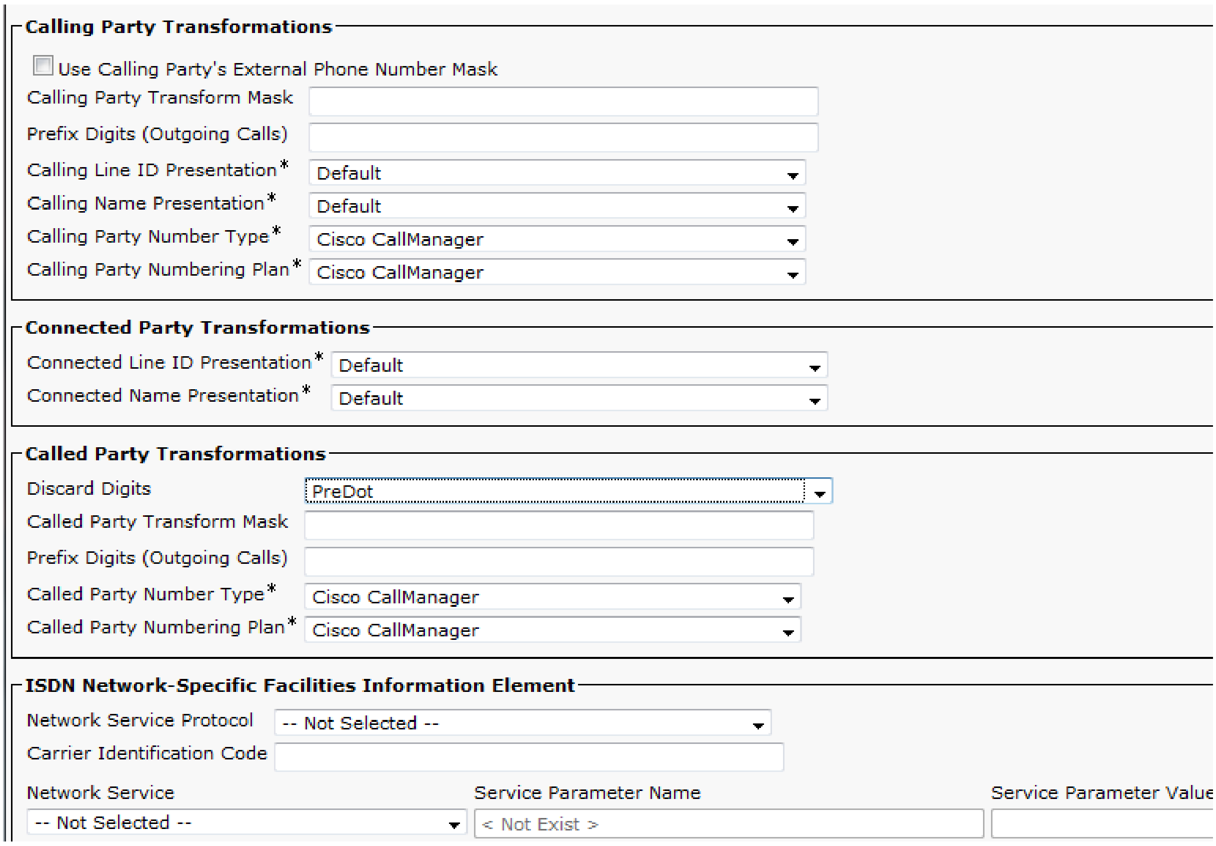
724! means prefix 724 and all other digits after

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 Transofrmation must be set up PreDot for the stripping of the 0.



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



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