



2N®

# VoiceBlue Next



## **2N® 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:

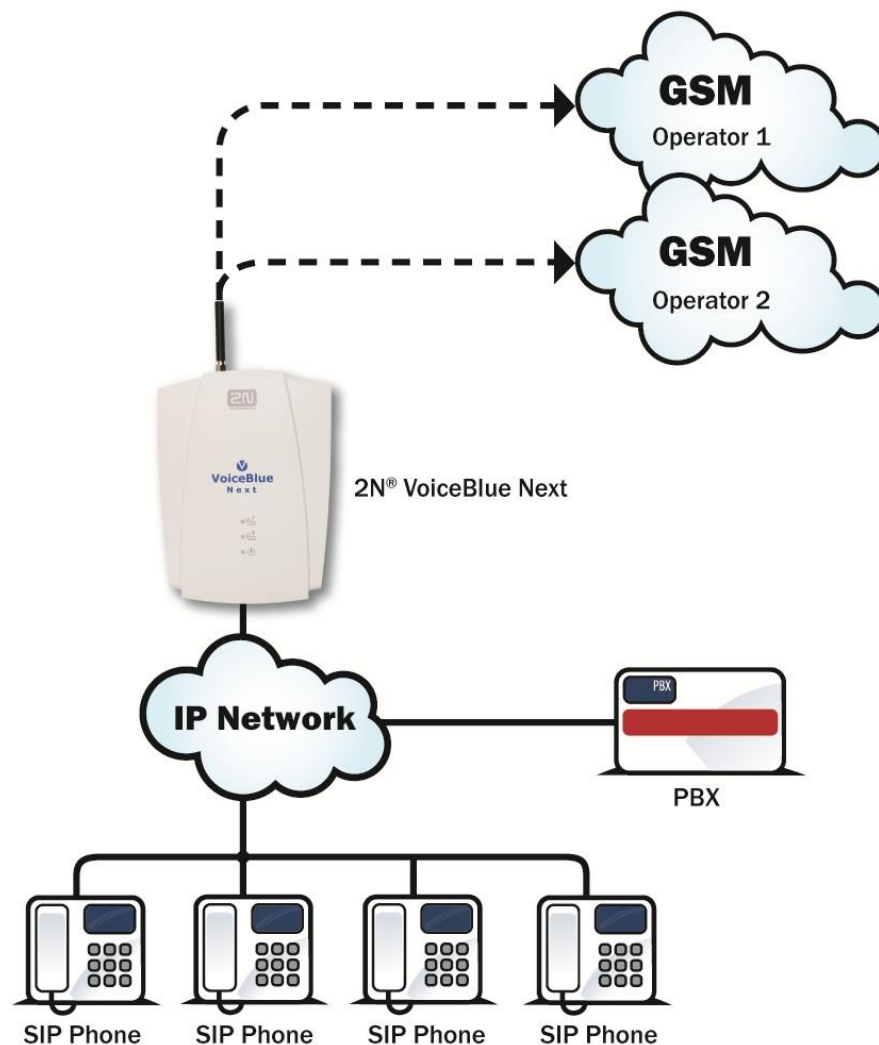
- IP address 192.168.22.42
- Incoming port: 5060
- Firmware: 01.00.04

### 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 displays the 2N Gateway configuration web interface. The left sidebar shows a navigation menu with 'Gateway configuration' expanded, listing various settings like System parameters, VoIP parameters, GSM basic parameters, etc. The main content area is titled 'Gateway' and includes buttons for 'Update' and 'Restart'. Under 'Gateway control', G711 and G729 codecs are set to '3 x 10ms'. The 'Codec priority' section shows three priority levels. The 'IP addresses' section contains the following configuration:

Parameter	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		

Two callout boxes provide context: one points to the 'SIP proxy (GSM→IP)' field with the text 'The IP address where the traffic is sent', and another points to the 'SIP proxy (IP→GSM)' field with the text 'The IP address and port which the traffic will come from'.

### 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.



 **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:**  
0. module  
1. module

**Outgoing:**  
1. Group  
2. Group

**Incoming:**  
1. Group  
1. 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)

The screenshot shows the 'Gateway' configuration interface. On the left is a sidebar with a '2N TELECOMMUNICATIONS' logo and a 'Gateway control' menu. The main 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: '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 is a section 'Send CLIP from VoIP to GSM/UMTS'. A red warning message at the bottom right states: 'Attention! Must be supported by your GSM / UMTS operator. In other case outgoing calls to GSM / UMTS can be rejected!'. The bottom of the interface has a 'Logout' button and some icons.

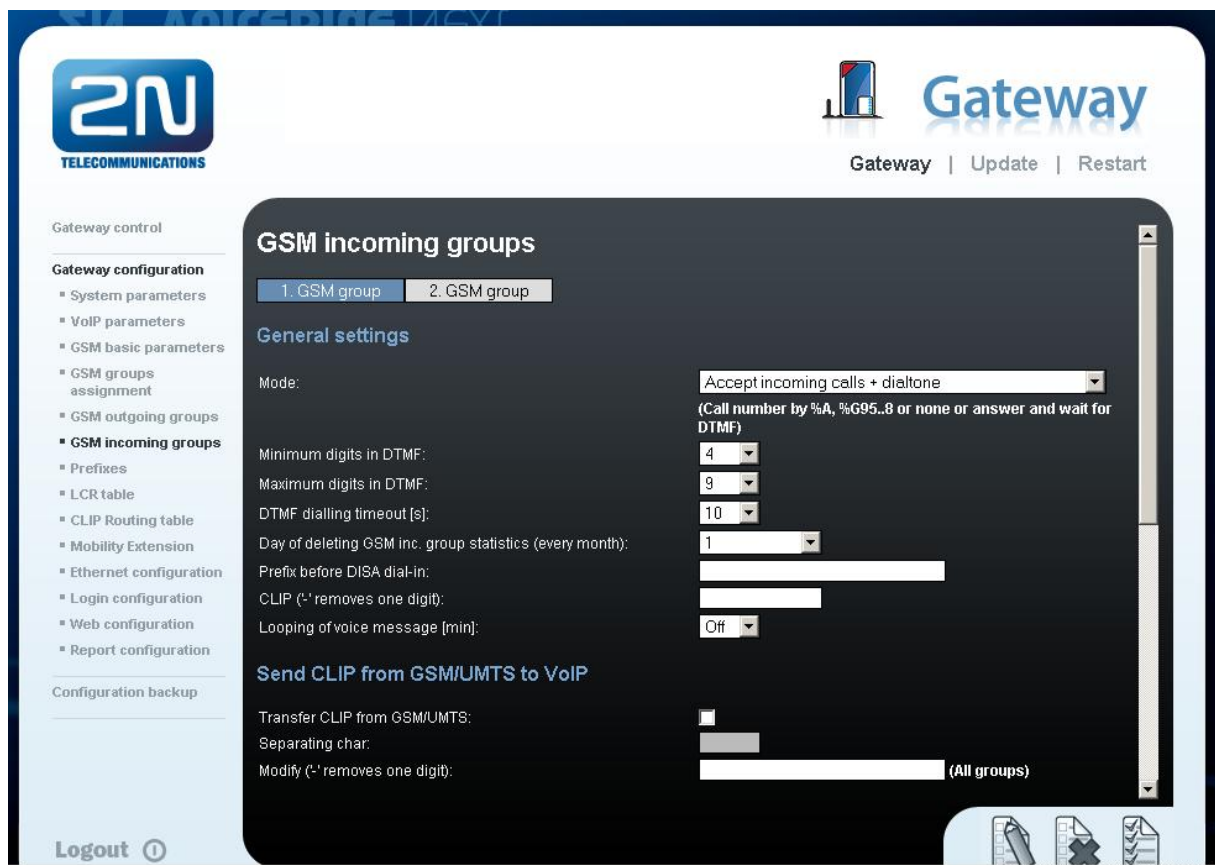
#### 5) Incoming calls

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.

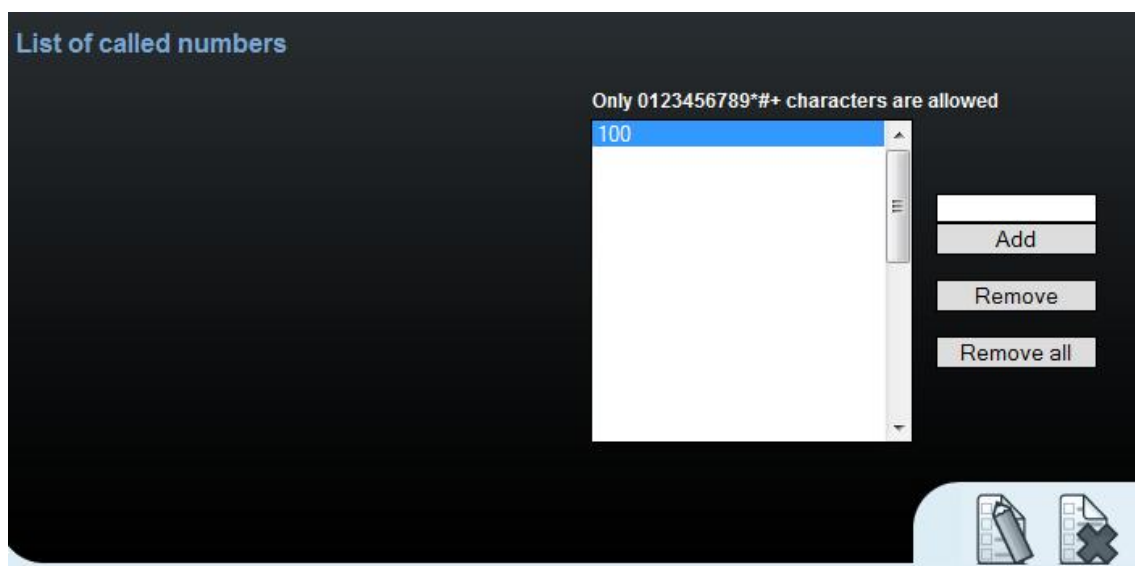
The screenshot shows a table titled 'GSM groups assignment'. It has three columns: 'Module:', 'Outgoing:', and 'Incoming:'. The 'Outgoing:' column has two dropdown menus, '1. Group' and '2. Group'. The 'Incoming:' column has two dropdown menus, both set to '1. Group'. The 'Incoming:' column is highlighted with a red rectangle.

Module:	Outgoing:	Incoming:
0. module	1. Group	1. Group
1. module	2. Group	1. Group

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).

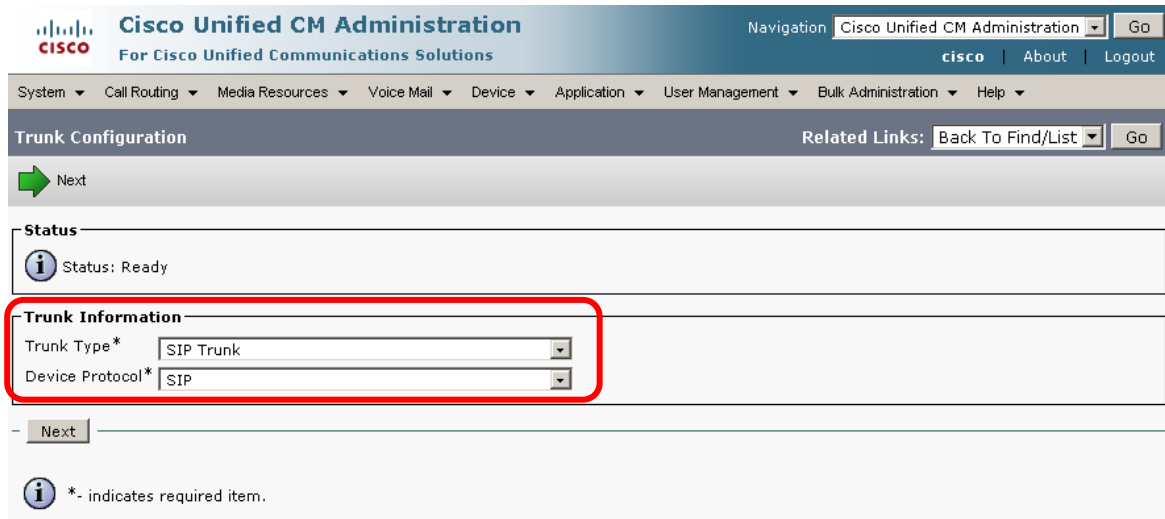


# CISCO CALL MANAGER SETTING

## 1) Create a new trunk

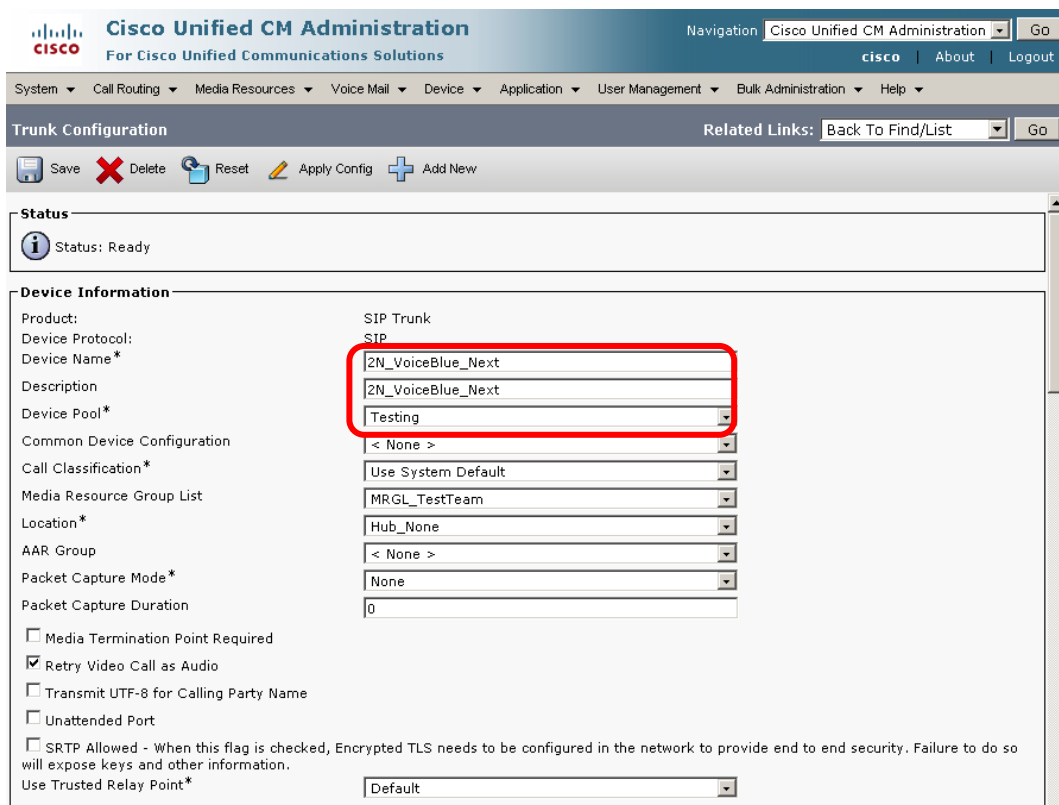
Add new trunk in the menu Device → Trunk → Add new

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



The screenshot shows the Cisco Unified CM Administration interface. The navigation bar includes System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main section is titled "Trunk Configuration" with a "Next" button. Below this, the "Status" section shows "Status: Ready". The "Trunk Information" section is highlighted with a red box and contains two dropdown menus: "Trunk Type\*" set to "SIP Trunk" and "Device Protocol\*" set to "SIP". A "Next" button is located below the "Trunk Information" section. A note at the bottom states: "\*- indicates required item."

Set up the trunk as in the picture below



The screenshot shows the Cisco Unified CM Administration interface. The navigation bar includes System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main section is titled "Trunk Configuration" with a "Related Links: Back To Find/List" and a "Go" button. Below this, the "Status" section shows "Status: Ready". The "Device Information" section is highlighted with a red box and contains several fields and dropdown menus: "Product:" set to "SIP Trunk", "Device Protocol:" set to "SIP", "Device Name\*" set to "2N\_VoiceBlue\_Next", "Description" set to "2N\_VoiceBlue\_Next", "Device Pool\*" set to "Testing", "Common Device Configuration" set to "< None >", "Call Classification\*" set to "Use System Default", "Media Resource Group List" set to "MRGL\_TestTeam", "Location\*" set to "Hub\_None", "AAR Group" set to "< None >", "Packet Capture Mode\*" set to "None", "Packet Capture Duration" set to "0", "Media Termination Point Required" (unchecked), "Retry Video Call as Audio" (checked), "Transmit UTF-8 for Calling Party Name" (unchecked), "Unattended Port" (unchecked), "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." (unchecked), and "Use Trusted Relay Point\*" set to "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.

[Clear Prefix Settings](#) [Default Prefix Settings](#)

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

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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**  
Info Status: Ready

**SIP Trunk Security Profile Information**

Name\*

Description

Device Security Mode

Incoming Transport Type\*

Outgoing Transport Type

☐ Enable Digest Authentication

Nonce Validity Time (mins)\*

X.509 Subject Name

Incoming Port\*

☐ 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](#)

Save
 Delete
 Copy
 Add New

**Status**

Status: Ready

**Pattern Definition**

Route Pattern*	<input type="text" value="0.6"/>
Route Partition	<input type="text" value="TestTeam"/>
Description	<input type="text" value="voice blue next"/>
Numbering Plan	<input type="text" value="-- Not Selected --"/>
Route Filter	<input type="text" value=" &lt; None &gt;"/>
MLPP Precedence*	<input type="text" value="Standard"/>
Resource Priority Namespace Network Domain	<input type="text" value=" &lt; None &gt;"/>
Route Class*	<input type="text" value="Standard"/>
Gateway/Route List*	<input type="text" value="TestTeam"/> <a href="#">(Edit)</a>
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <input type="text" value="Kein Fehler"/>

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\*

**Connected Party Transformations**

Connected Line ID Presentation\*

Connected Name Presentation\*

**Called Party Transformations**

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type\*

Called Party Numbering Plan\*

**ISDN Network-Specific Facilities Information Element**

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value=" &lt; Not Exist &gt;"/>	<input type="text"/>



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