

Configuring Satelco VoIP – GSM gateways with Cisco call manager

The present document is a step-by-step guide for configuring the 2N VoIP GSM gateways VoiceBlue and VoiceBLue Enterprise and Cisco Call Manager software IP PBX. It includes essential steps for interconnecting the two products rather than a detailed configuration analysis of all units.

What is VoiceBlue Lite?

The Satelco - VoiceBlue GSM gateway is an ideal complementary product to any SIP-based IP PBX. It's suitable for small and medium companies with IP infrastructure and for companies with international affiliates. With VoiceBlue you gain significant savings on outgoing and incoming calls from IP to GSM networks and backwards. Thanks to the efficient and powerful Least Cost Router (LCR), the VoiceBlue GSM gateway always chooses the cheapest possible way to route the call (according to GSM prefixes, free minutes on SIM cards etc.). Using the VoiceBlue gateway you achieve complete independence when connecting to GSM networks and maintain 100% control of your GSM call costs. With the help of voice prompts and efficient Dynamic Clip Routing, VoiceBlue routes incoming GSM calls to the right IP phones.

What is Satelco VoiceBlue Enterprise?

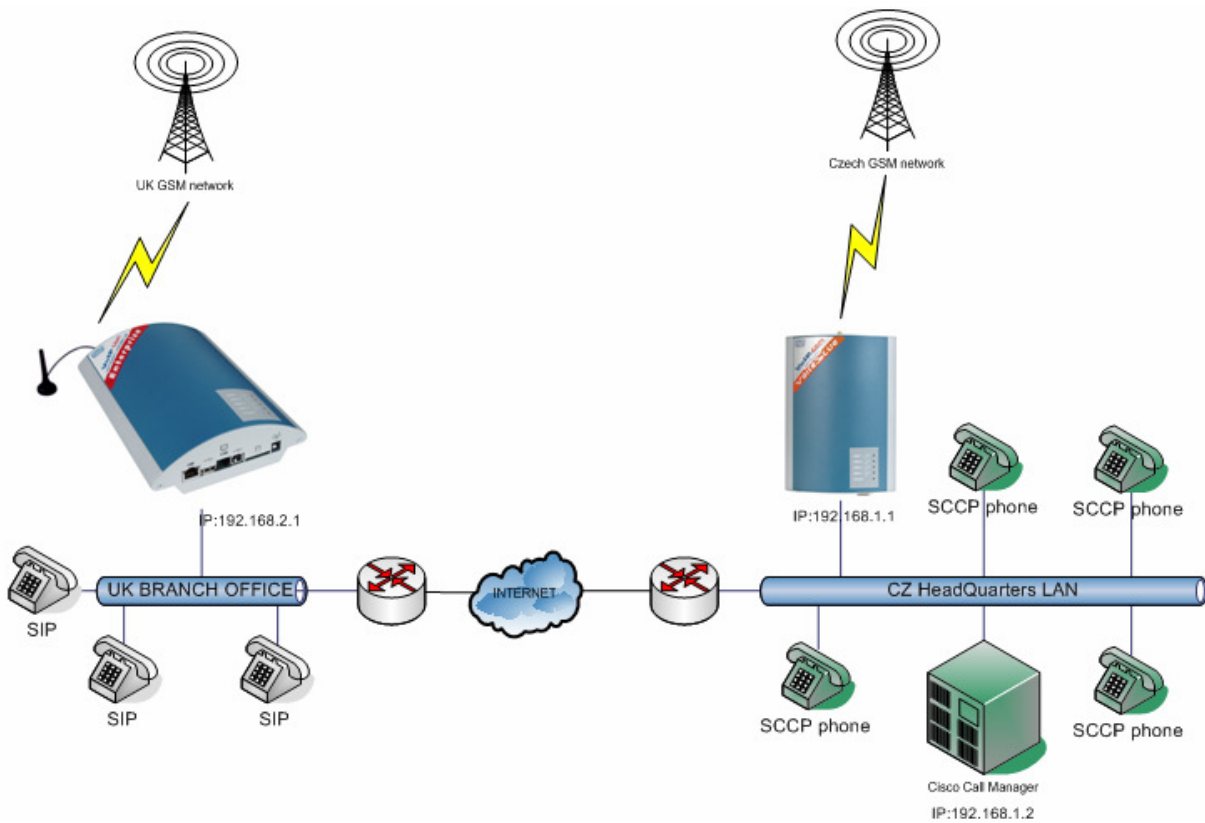
VoiceBlue Enterprise is a VoIP-to-GSM Gateway supporting SIP and H.323 at the same time. It is a complementary product to any SIP or H323 based IP PBX and can also be used as a substitute to any SIP based IP PBX because the SIP proxy server can be included.

The VoiceBlue Enterprise fixed cellular terminal can be used not only as a gateway to GSM but also a protocol translator between the SIP and H.323 environments (the scheme below). As VoiceBlue Enterprise uses the Siemens MC75 module, it is capable of data transfer (GPRS and EDGE technologies are implemented) and therefore can serve as a gateway from your LAN to the Internet. A firewall is embedded to provide maximum security. There is an option of SMS sending and receiving through a web or SMTP/POP3 interface. If an external USB hard disc is added, VoiceBlue can serve as an SMS server for SMEs as well.

Our scenario

In this article we are going to deal with configuring gateways and Cisco Call Manager system, connected in the way this scheme can resemble the practical application in the best possible way. Let's imagine a company using CCM as the main exchange in its headquarters in the Czech Republic and having one branch office in UK. Together with CCM, there is Satelco VoiceBlue gateway installed in the company headquarters, which will be used to complete calls to local GSM network.

The branch office in UK is so small it does not pay off to equip it with the classical exchange and most of its employees are out of their offices most of the time. This implies that most of the communication between the headquarters and the branch office employees will be performed over mobile phones in the UK GSM network. For that reason the branch office exchange is replaced with the Satelco VoiceBlue Enterprise system, which will ensure PBX functions for employees and at the same time the connection to GSM network, together with SIP telephones located in the UK branch office. Thanks to the Internet connection of the headquarters and the branch office the calls between companies can be routed free of charge and calls to distant GSM networks can be charged locally. (To make it simple let's suppose there is an IP VPN created between the headquarters and the branch office so the whole operation will function in a transparent way.)



Now we have the scenario, but the question is what part individual elements in our model will play. First an internal numbering plan has to be created. To make it simple let's say, we will use four-digit numbering and all telephone numbers in the headquarters will start with 1XXX and all telephone numbers in the branch office will then be numbers starting with 2XXX. Furthermore let's say that all calls to numbers with prefixes 004206,004207,004208,6,7,8 will be routed to VB IP address in the company headquarters, all calls starting with 00420 will be routed to CCM and calls with 0044 will be routed to VoiceBlue Enterprise. This way it is ensured that if somebody in the branch office calls the Czech number, the VoiceBlue Enterprise will send the call to CCM and vice versa.

In accordance with the previous it is possible to create a simple routing table for each device.

Prefix	Number of digits	Destination IP address
2	4	192.168.2.1
004206	13	192.168.1.1
004207	13	192.168.1.1
004208	13	192.168.1.1
6	9	192.168.1.1
7	9	192.168.1.1
8	9	192.168.1.1

Tab. x.x: Routing table in CCM

Prefix	Number of digits	Destination IP address
1	4	192.168.1.2
00420	13	192.168.1.2
0044	13	local gateway

Tab. x.x: Routing table in VoiceBlue Enterprise

Configuration

CCM

We will start with what seems to be the most difficult, that is with CCM. We will not deal with the telephone installation and CCM configuration for communicating with them here ... it is for an independent document. We are going to focus only on SIP Trunk configuration and routing rules creation.

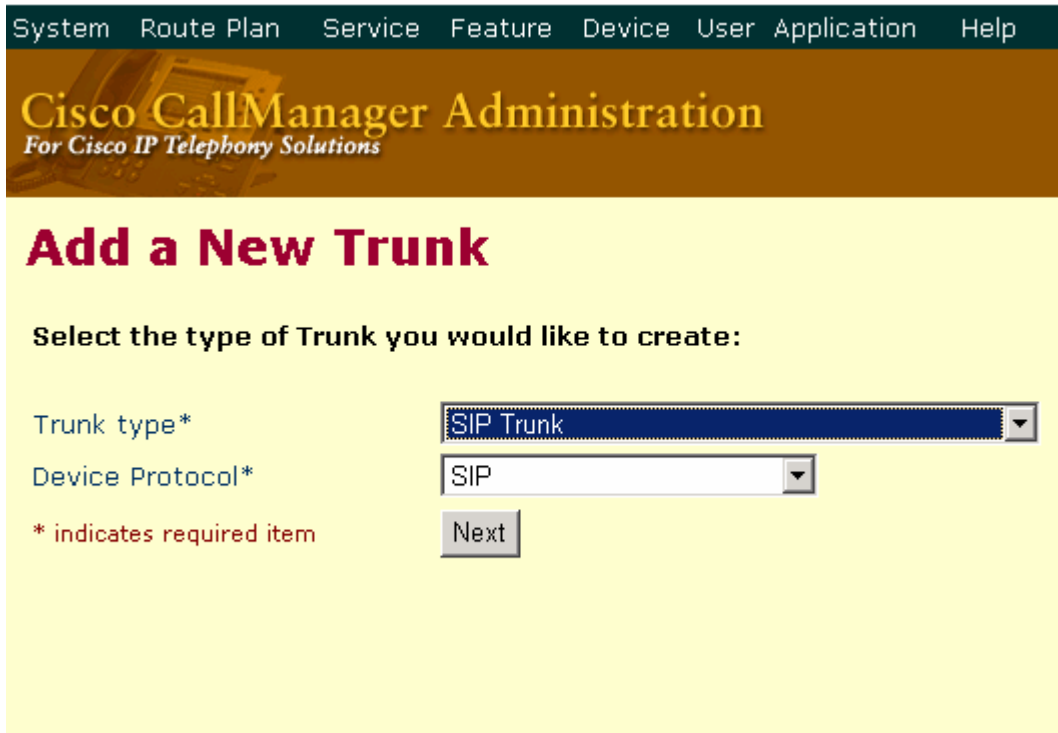
SIP Trunk creation

SIP Trunk is a kind of virtual interface, which points at the IP address of a distant device. This interface only knows where its distant end is and what protocol it can use to communicate with it.

From the Main Menu select Device > Trunk



click on Add new trunk and select protocols.



System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration

For Cisco IP Telephony Solutions

Add a New Trunk

Select the type of Trunk you would like to create:

Trunk type*

Device Protocol*

* indicates required item

After clicking on Next, Trunk parameters are displayed.
Change parameters according to pictures and click on INSERT.



System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration

For Cisco IP Telephony Solutions

Trunk Configuration

[Add](#)
[Back to F](#)
[Depenc](#)

Product: SIP Trunk
Device Protocol: SIP
Status: Ready

Device Information

Device Name*

Description

Device Pool*

Media Resource Group List

Location

AAR Group

Media Termination Point Required

Destination Address*

Destination Address is an SRV

This way an interface for VoiceBlue in headquarters has been created.

In the same way create a trunk for VoiceBlue Enterprise, so finally the trunk list looks the same as in the following picture. To display data click on Find.



System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration
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CISCO SYSTEMS

Find and List Trunks [Add a New Trunk](#)

2 matching record(s) for Device Name begins with ""

Find trunks where begins with

Show items per page.

To list all items, click Find without any search text, or use "Device Name is not empty" as the search criteria.

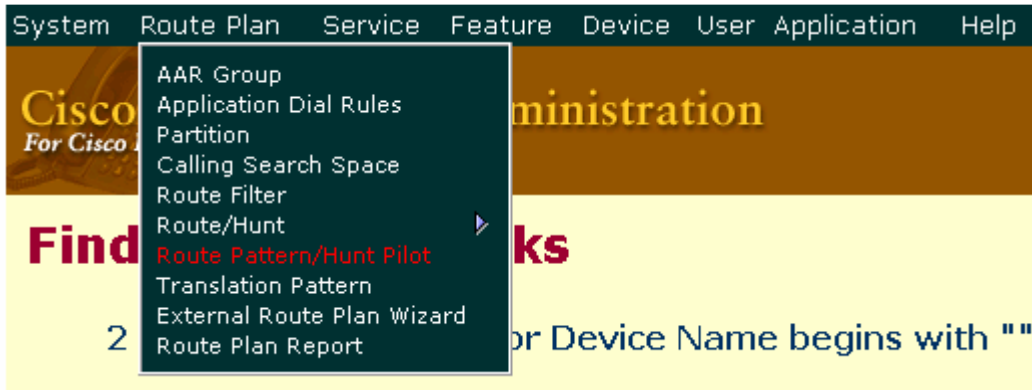
Matching record(s) 1 to 2 of 2

<input type="checkbox"/>	Device Name	Description	Device Pool
<input type="checkbox"/>	 VoiceBlue	VoIP gsm gateway in CZ	Default
<input type="checkbox"/>	 VoiceBlue_Enterprise	VoiceBlue_Enterprise in UK	Default

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Creation of a routing table

From the Menu select Route Plan > Route Pattern/Hunt Pilot



System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration

Find and List Trunks

2 matching record(s) for Device Name begins with ""

- AAR Group
- Application Dial Rules
- Partition
- Calling Search Space
- Route Filter
- Route/Hunt
- Route Pattern/Hunt Pilot**
- Translation Pattern
- External Route Plan Wizard
- Route Plan Report

Click on Add new route

Create the first route pattern according to our routing table for prefix 2 and a total number of digits 4. To make decisions CCM uses the algorithm similar to regular expressions. In our case a single expression, which is X and indicates any number, will do. So the expression in our case will look as follows: 2XXX.

System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration

For Cisco IP Telephony Solutions

Route Pattern/Hunt Pilot Configuration

[Add a](#)
[Back to Find/List F](#)

Route Pattern/Hunt Pilot: 2XXX

Status: Insert completed
Note: Any update to this Route Pattern or Hunt Pilot automatically resets the associated ga

Pattern Definition

Route Pattern/Hunt Pilot*

Partition

Description

Numbering Plan*

Route Filter

MLPP Precedence

Gateway or Route/Hunt List* (Edit)

Set values according to the picture and click on Insert.
Now create the rest of route patterns according to our routing table in order the route pattern printout can look as follows:

Matching record(s) 1 to 7 of 7

<input type="checkbox"/>	Route Pattern/Hunt Pilot	Partition	Description	Route Filter	Gateway/Route List	Copy
<input type="checkbox"/>	004206XXXXXXXX				VoiceBlue	
<input type="checkbox"/>	004207XXXXXXXX				VoiceBlue	
<input type="checkbox"/>	004208XXXXXXXX				VoiceBlue	
<input type="checkbox"/>	2XXX				VoiceBlue_Enterprise	
<input type="checkbox"/>	6XXXXXXXXX				VoiceBlue	
<input type="checkbox"/>	7XXXXXXXXX				VoiceBlue	
<input type="checkbox"/>	8XXXXXXXXX				VoiceBlue	

Configuring Satelco VoiceBlue Enterprise

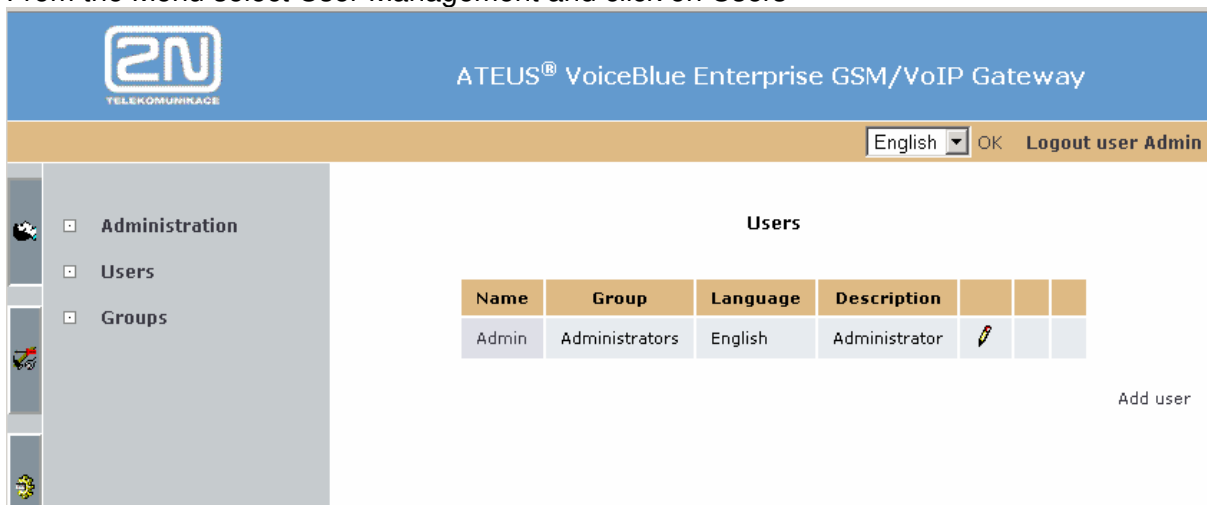
Now the headquarters has been configured. CCM knows which way to send calls to the Czech GSM networks and which way to the English ones. At the same time it also knows when somebody calls an internal line starting with 2, this will also be routed to England. So we need to perform the configuration of the whole branch office in England, i.e. VoiceBlue Enterprise.



Configuring internal SIP telephones

Let's start with creation of environment for the branch office internal telephones. Let's suppose there are 4 SIP telephones in the branch office and numbers 2001, 2002, 2003 and 2004 will be assigned to them. Let's have a look how to do it right now.

Set first the IP-Address of Voiceblue Enterprise by using the Communication-Port (see also Manual on chapter 3.1 to 3.2, then you can start access via Web-Browser.

From the Menu select User Management and click on Users



Name	Group	Language	Description			
Admin	Administrators	English	Administrator			

Add user

After adding the user, continue on Add user and proceed according to the following picture. Here Username represents the telephone number of the given telephone and Password is the password which SIP telephone will use to connect to VoiceBlue Enterprise.

Continue in the same way also for the rest of phones so that finally the Users Table looks as follows:

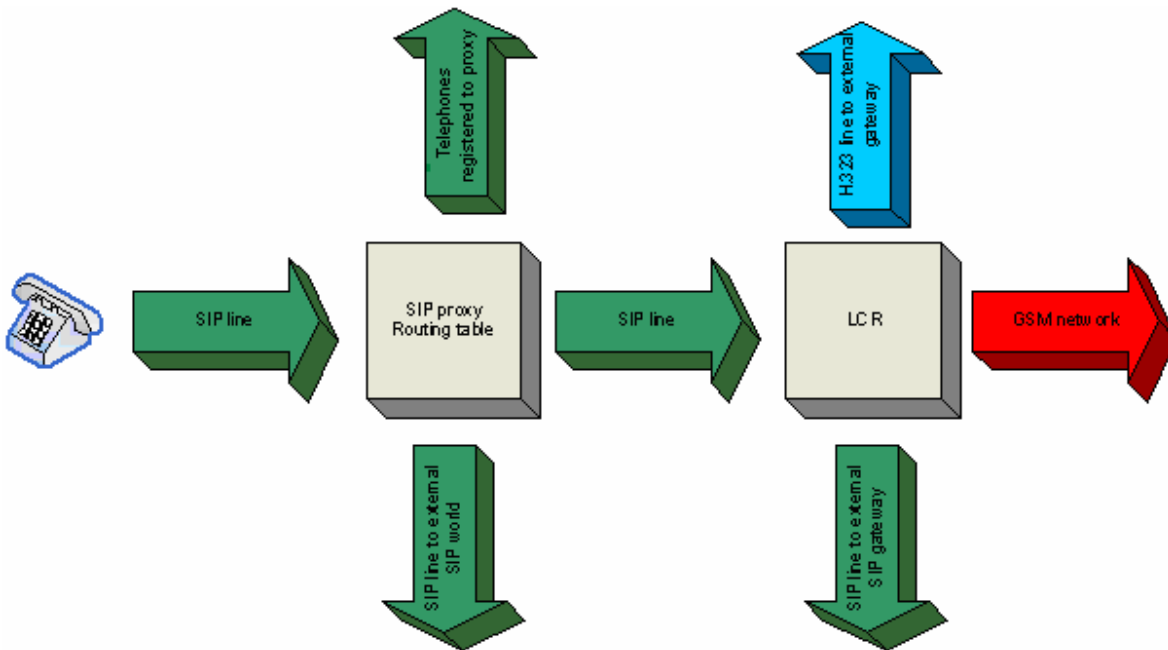
Name	Group	Language	Description			
2001	Administrators	English				<input type="checkbox"/>
2002	Administrators	English				<input type="checkbox"/>
2003	Administrators	English				<input type="checkbox"/>
2004	Administrators	English				<input type="checkbox"/>
Admin	Administrators	English	Administrator			

Now you can register SIP telephones to this IP GSM gateway easily by setting VBE IP address in the SIP proxy field and by signing it in by the means of accounts we have just created. After finishing this action it is possible for telephones in the branch office to make phone calls among each other.

Routing tables

The internal SIP proxy has been configured in order to register SIP telephones and enable to make phone calls among each other.

Finally, let's move to the implementation of our routing table. Before creation of routing rules it is important to realize the way the routing functions inside the VBE. Basically, VBE has two routing tables, first of them being marked as SIP proxy Routing table and the second as LCR. The sequence of decision rules is indicated in the following picture.



So when the telephone registered to the internal SIP proxy wants to dial a random number, it sends the INVITE packet to VBE or to its internal SIP proxy server. It is processed in SIP proxy rules and then it is either transmitted directly to other SIP gateway or it is transmitted to the internal LCR in case we need to route the call either to GSM or to H.323 networks.

SIP proxy

So first create routing rules in SIP proxy table. To configure a SIP proxy table click on Services > SIP proxy.

Here create two routes, one to the internal LCR table to route calls to GSM and one to CCM in the company headquarters.

So the call with prefix 0044 will be routed to LCR through the internal SIP line.

If prefix:	<input type="text" value="sip:0044"/>
Strip:	<input type="text" value="0"/>
Add:	<input type="text"/>
Do action:	<input type="text" value="connect to VoiceBlue"/>
With parameter:	<input type="text" value="SIP-14"/>








The call to the headquarters will be transmitted to Call Manager IP address.

If prefix:	<input type="text" value="sip:00420"/>
Strip:	<input type="text" value="0"/>
Add:	<input type="text"/>
Do action:	<input type="text" value="rewrite host"/>
With parameter:	<input type="text" value="192.168.1.2"/>

And so will all calls from the range of local numbers in the headquarters.

If prefix:	<input type="text" value="sip:1"/>
Strip:	<input type="text" value="0"/>
Add:	<input type="text"/>
Do action:	<input type="text" value="rewrite host"/>
With parameter:	<input type="text" value="192.168.1.2"/>

After all settings the routing table in SIP proxy should assumedly look as follows:

If prefix	Strip	Add	Do action	With parameter			
sip:0044	0		connect to VoiceBlue	SIP - 14			<input type="checkbox"/>
sip:00420	0		rewrite host	192.168.1.2			<input type="checkbox"/>
sip:1	0		rewrite host	192.168.1.2			<input type="checkbox"/>
else	0		lookup registration				

LCR table

Now LCR table has to be adjusted. Before defining a routing table it is necessary to create the route for this direction. In VEB the routes to GSM network are represented by GSM group. To make it simple, let's suppose we have SIM cards with the same tariff from one operator and therefore only one GSM group is to be created. See VBE manual for more complex configurations.

Basic | Limits

GSM group name:

Lines of GSM group:
 GSM module - 1
 GSM module - 2
 GSM module - 3
 GSM module - 4

Time intervals:
 weekdays
 weekend
 workdays

CLIP/CLIR:

Roaming enabled:

Roaming network number:

Description:



Buttons: Add SIP line, Add H.323 line, Add SIM card, Add day group, Add time range

To create GSM group proceed as follows:

From the Menu select Configuration > LCR > GSM group

Name the group at your discretion and mark all GSM modules you want to use.

After adding the group there should be the following printout in the GSM groups list.

GSM group name	Path lines	Time intervals	Description			
To GSM network	GSM module - 1 GSM module - 2 GSM module - 3 GSM module - 4	weekdays weekend workdays				<input type="checkbox"/>

Now there are no obstacles to create a LCR table. In LCR section click on ADD button and proceed according to pictures:

Enabled:

Destination name:

Prefix 1:

Description:

The prefix box contains the regular expression, which is suitable for any number starting with 0044.

GSM group name:

Priority type 1:

Priority type 2:

Priority type 3:

Priority type 4:

Description:

In the next step a single outgoing group, which has been defined, will be selected automatically and therefore it is enough to enter by clicking on Add button. Your LCR table should look as follows:

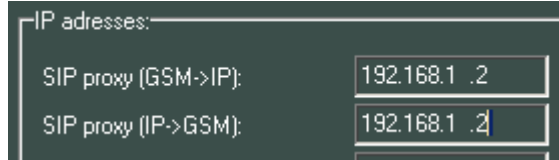
Destination	<input type="checkbox"/>	Enabled	GSM groups	Description
To local GSM	<input type="checkbox"/>	<input checked="" type="checkbox"/>	To GSM network	

This way the configuration in the branch office is completed and we should be able to get connected to headquarters' local numbers and to UK GSM networks both from the UK branch office and from the headquarters.

Configuring VoiceBlue Lite

In this section you can find the most basic steps to configure VoiceBlue Lite. For more information see the previous articles or the manual.

In order VB can accept calls routed to it from CCM we have to insert CCM IP address to the SIP proxy field in ethernet parameters section.

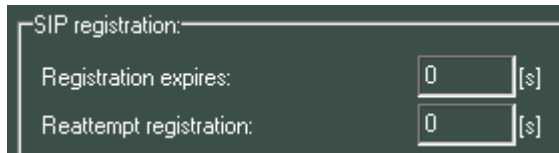


IP addresses:

SIP proxy (GSM->IP): 192.168.1 .2

SIP proxy (IP->GSM): 192.168.1 .2

At the same time forbid the registration by inserting 0 to the fields Registration expires and Reattempt registration.



SIP registration:

Registration expires: 0 [s]

Reattempt registration: 0 [s]

Create a network list for numbers starting with prefix 00420.

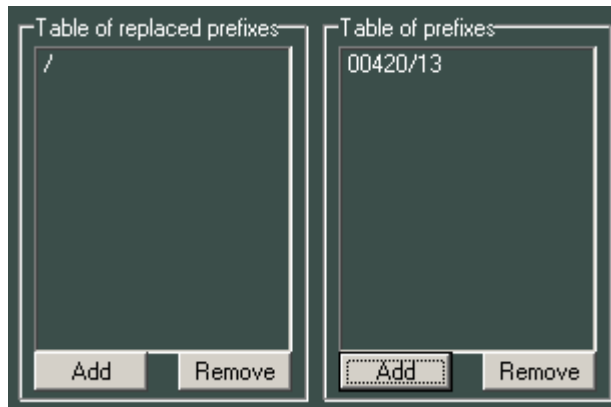


Table of replaced prefixes

/

Add Remove

Table of prefixes

00420/13

Add Remove

And finally create add record to LCR table.

Network	Using time	Groups	Limit
1	00:00/24:00	1	0

At that moment everything should function according to our scenario. However, in case you face any trouble, go through the whole configuration making sure you did not forget about anything and if you do not eliminate the trouble you can always contact satelco@satelco.ch which is free of charge for all Satelco customers.