# OpenScape Business

# How to Configure SIP Trunk for BroadCloud - USA

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# Table of History

| Date | Version | Changes |
| --- | --- | --- |
| 06.04.2017 | 1.0 | Released |
| 09.09.2024 | 1.1 | editorial changes |
|  |  |  |

**Note**: The basis for this document is the current OpenScape Business at the time of certification. Since OpenScape Business is constantly developed, input masks and interfaces as well as requirements may change in the future. The settings and entries described here then apply accordingly.

# Information

The BroadCloud provider is available and released for the countries USA, UK, Germany, France, Italy and Australia.

**This configuration guide is valid for USA only.**

## Trunk Configuration Data provided by BroadCloud

The configuration data needed to setup the SIP trunk can be found on the BroadCloud Enterprise Portal under SIP Trunking Page > Trunk Group > Edit under the selected Trunk Group:
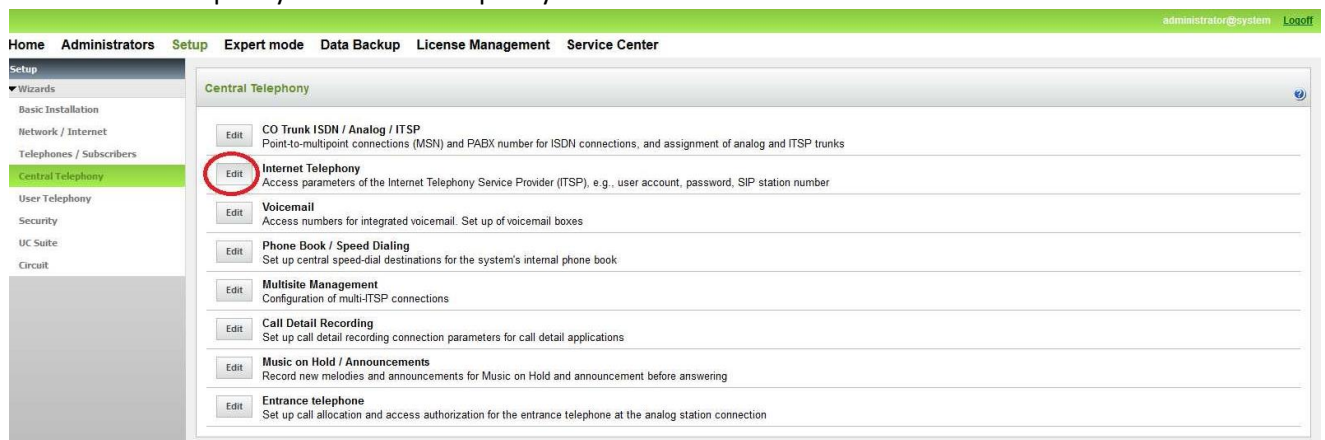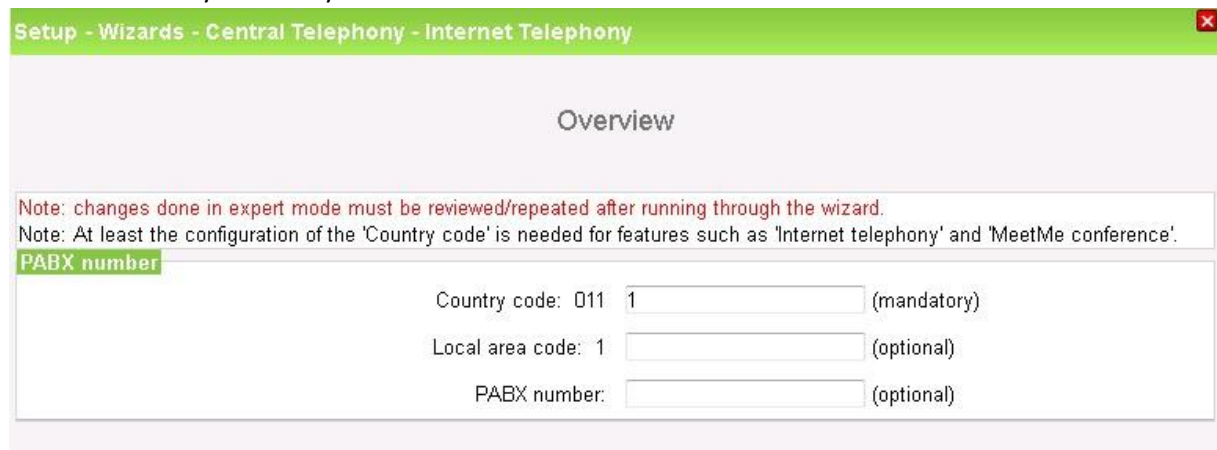
# Configuration Wizard

## Internet Telephony

Go to Central Telephony – "Internet Telephony"



The overview page appears for entering the location data. The most flexible type of configuration is to enter the Country code only.



Click [OK & Next].

Provider configuration and activation for Internet Telephony -> No call via Internet -> uncheck Use County specific view: United States of America and select "BroadCloud USA".

Activate Provider and click on [Edit].

On the next page you have to enter the following information:

- **Domain Name**
  The SIP Domain Name can be found on the BroadCloud Enterprise Portal, under SIP Trunking Page > Trunk Group > Edit under the selected Trunk Group. The part required is found on the section called Registered Domain.

  !!! The Domain in the screenshot is only an example. !!!



- **Provider Registrar** and the **Provider Proxy**. Use the country individual names:

  **BroadCloud USA Zone 1:**   chcgoutpostsipconnect-udp.broadcloudpbx.com
  **BroadCloud USA Zone 2:**   dllsoutpostsipconnect-udp.broadcloudpbx.com
  The **Provider Outbound Proxy** is not used and therefore left deactivated.

Click [OK & Next].

In this dialog the specific customer SIP Userdata will be configured.



Click on [Add].

Data provided on the BroadCloud Enterprise Portal, under SIP Trunking Page > Trunk Group > Edit under the selected Trunk Group:



The data provided on this page are used to fill the following fields:

**Internet telephony station**: **Username** is inserted here (e.g: 8324583770)
**Authorization name**: **Username** is inserted here (e.g: 8324583770)
**Password**: **Password** provided by BroadCloud
**Default number**: Main number of connection. The default number is used as outgoing number when no DDI number is assigned to a station. (e.g: 8324583770). Usually the **Lead Number** is entered here.

Enter the relevant data and click [OK & Next].
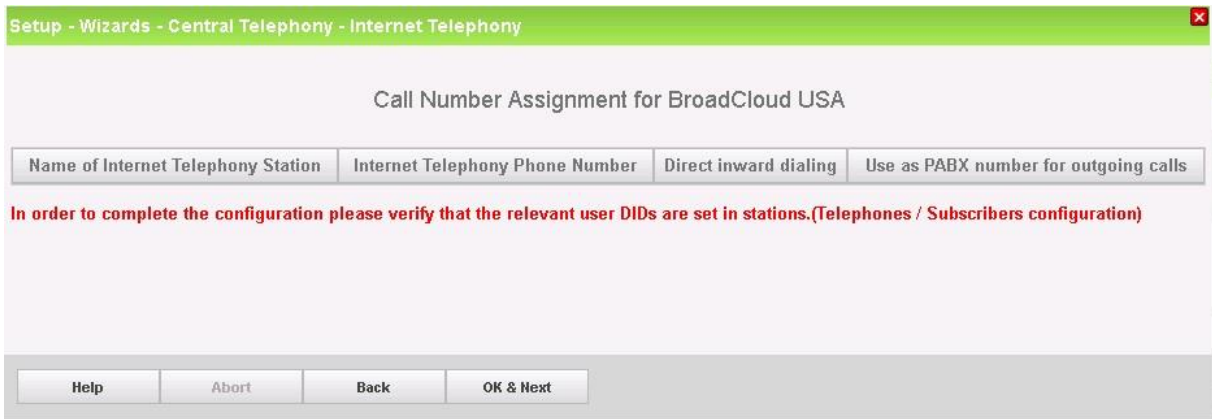


Click [OK & Next]

Click [OK & Next] (no input needed)
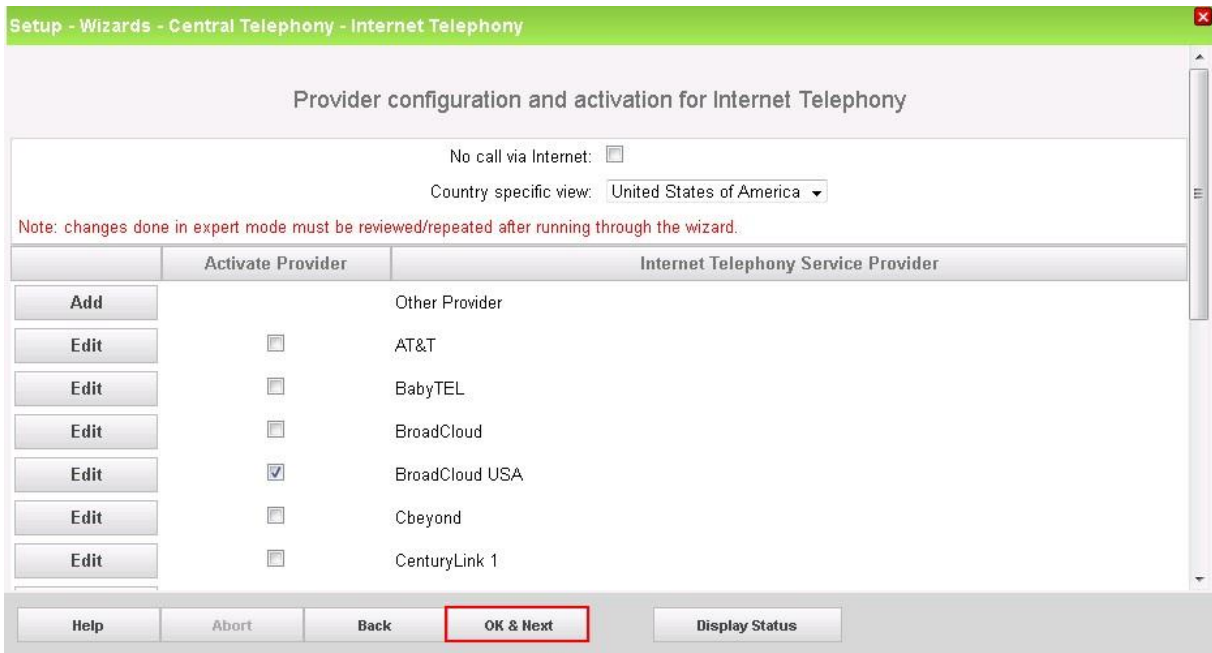


Click [OK & Next]

# Define bandwidth (# Trunks)

The amount of simultaneous Internet (**Assigned Lines**) calls must be aligned with the **Maximum Active Calls** assigned to the Trunk Group on the BroadCloud Enterprise Portal.





Click [OK & Next]

## Special phone numbers

In this dialog it is possible to route special phone numbers.



Click [OK & Next]

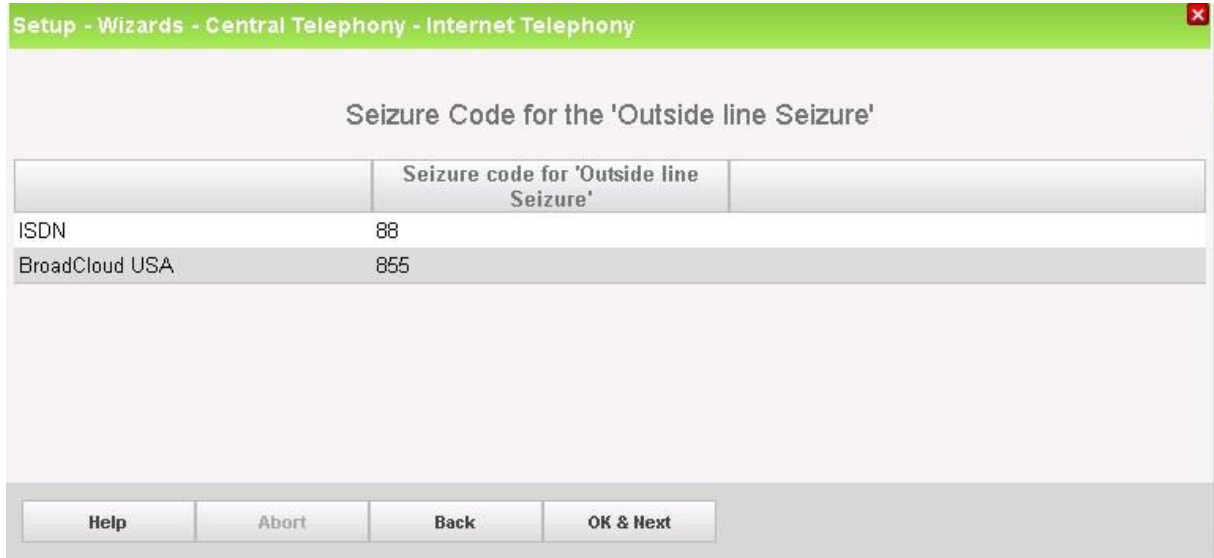On next page status of ITSP is displayed.



Click [Next]

„Exchange Line Seizure":

Select which trunk will access code 9. Enter the local area code without prefix digits (needed only when local area code was not entered in first step PBX number)



Click [OK & Next]

Overview with all configured „Outside line Seizure" are displayed.



Click [OK & Next] and on the next page [Finish]

# DID configuration

In the DID Section, the full DID will need to be entered without the country code.

All numbers shown in the BroadCloud **Trunk user DDI** can be used as DID and can be found on the BroadCloud Enterprise Portal under the SIP Trunking Page > Trunk Group > **Assigned** in the column **Phone Number**:

# Additional manual Configuration

## License
Add the "S2M/SIP Trunk" license to the SIP-Trunk



## Route configuration (optional)
The route configuration will be created automatically. It should look like below.
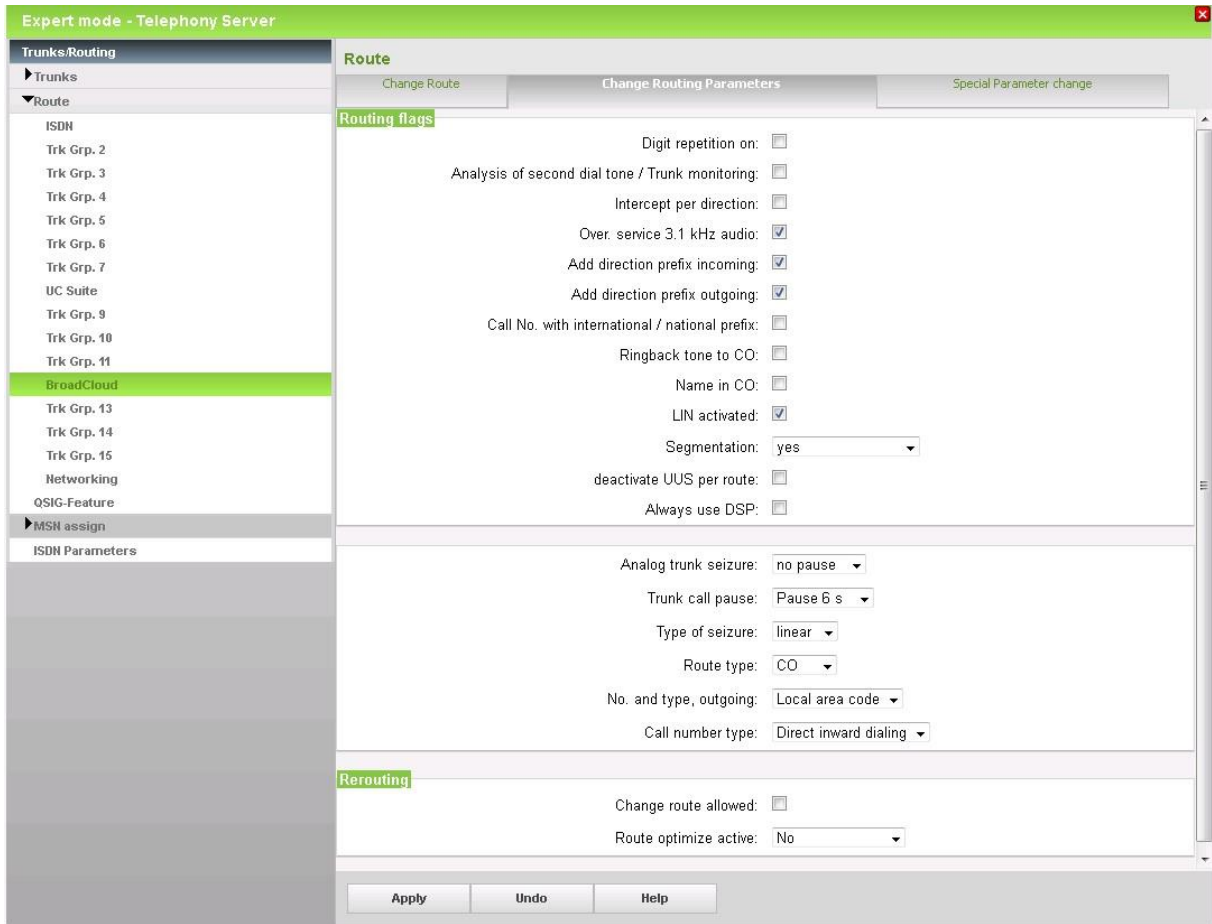
Best praxis is to enter the default Access Code in the field "Seizure code", enter the Country code for the PABX number-incoming and select Location number.



The route parameters:

## LCR  changes ( optional)

In V2R2 all necessary LCR rule will be created automatically. It should look like below.

In expert mode , Telephony Server, LCR , - change the "Routing Table" in the way that **all Dial plans related to SIP (BroadCloud) route should follow the Routing Table 4**.

If not changed manually, the Routing table 4 uses the Dial Rule 'SIP' which by default has the Dial Rule format= 'A', Network Access='Main Network Supplier' and Type='Unknown' .

The dial rule manipulates the number so that the called party number is national for national/local calls and international for international calls.

Routing table 4:



Dial rules:

# How to check the status of SIP connection with BroadCloud

"Status" provides information about the registration status of the ITSP. The ITSP registration can also be restarted here.



# Known restrictions:

1. CLIP No Screening is not supported for outgoing calls. A phone cannot present a number which is not owned by the provider trunk.
2. COLP: Provider doesn't support Connected Line Identification Presentation
3. Special scenario in case of non-DID phone and forwarded attendant phone.
   Description in detail: A non-DID Phone usually present the attendant DID for external calls. In case the attendant activates call-forwarding to an external destination and the non-DID phone is calling the attendant, the call will be rejected by the provider. Reason: The provider does not accept calls with the same FROM and DIVERSION number.

# Appendix

## Supported Features

1. The Provider use dynamic Registration
2. The Provider support codec G.711, G.729 and G.722. But it is not allowed to use G.722 only
3. Fax: T.38 is supported by the provider.
4. Diverted calls present the number of the caller:
   In a transit call forwarding scenario the provider provides the number of the caller.
   e.g. external A calls internal B. B is forwarded to external C. C get the number information of A
5. MultiSite Scenarios is supported: The Provider is able to provided numbers in different Areas in the same country at the same sip-trunk.
6. DTMF: RFC2833 out-of-band is supported

## Supported Numbering Formats

**US Provider**

Outgoing call

Called Party (REQUEST, TO):

National: 9193789301@<SP FQDN>

National with prefix: **1**9193789301@<SP FQDN>

International with prefix: **011**498944234199905@<SP FQDN> Calling
Party (FROM, PAI, Diversion):

National: 8324583770@<SP FQDN>


Incoming call

Called Party (REQUEST, <u>TO</u>):

National: 9193789301@<SP FQDN>

International with prefix: +49894423410@<SP FQDN> Calling
Party (<u>FROM</u>):

National: 8324583770@<SP FQDN>