Basic Configuration of the Subscriber

Introduction
Here several features and supplementary services of the subscriber can be configured and/or activated/deactivated:

◊ Defining the identity of the subscriber, for example, which public telephone number (CLIP) shall be displayed or if an anonymous (CLIR) call shall be placed.
◊ Display of the SIP Login Credentials
◊ Defining the language to be used in standard announcements or in the AdminCenter Web GUI
◊ Defining the behavior of the call waiting CW if the user is busy with another call
◊ Defining a vPBX department for the telephone number
◊ Defining the location for emergency calls
◊ Defining IP subnet(s) or a SIP Profile which contains a list allowed IP subnets where VoIP devices are allowed to register to this user account
◊ Defining whether the user has an account at the "UCC WebRTC Conference Service" .
◊ Defining whether conversations are to be recorded and the audio file sent to a predefined email address

Note	It is possible that the provider did not enable all features!

Contents

The Subscriber Identity

General information about the identity of the subscriber can be transferred to the called party. This information can be displayed on the called side, depending on the capabilities of the terminal.

The following settings are possible in order to influence the display of the own identity:

◊ A "Name" can be defined for the telephone number of the subscriber, e.g. the personal name or company name. This information is used as a description of the telephone number and is transmitted as "Display Name" to the called party.

> Information for vPBX subscribers:
> The name is transmitted only within the vPBX. The name is not transmitted with outbound calls to the public telephony.

◊ In "Displayed external Number" the own public telephone number can be defined, which is transmitted to the called party and eventually displayed.

> Information for vPBX subscribers:
> Here another public vPBX phone number can be selected for display
> Example:
> The members of the support team do not want to have displayed their direct public telephone number, but the official support telephone number of the company instead.

◊ In "Suppress own number" can be configured how the transmitted telephone number shall be displayed:

- "Do not change"
  The information provided by the telephone or vPBX telephone number of the caller is transferred unchanged
- "Yes"
  The telephone number is always suppressed (CLIR) anonymous call
- "No"
  The public telephone number, which is used for the call is displayed (CLIP).

◊ In "Mobile Number" the associated mobile telephone number of the user can be configured. The user's mobile number is needed in various features, e.g. "Call Through" in the "an IP-Phone" mobile app.

SIP Credentials of the Subscriber Account

The SIP credentials "Domain", "Authenticating Name" and "Password" for the telephone number may be displayed and changed.

Note	The SIP credentials must be synchronized to the associated SIP devices!

Warning	• If not handled with care it is possible that the connectivity is not working correctly! • SIP devices using this SIP credentials must be reconfigured accordingly! • Configured Line Key on SIP telephones must be synchronized! • These values must be handled **confidential**. They open the possibility of misuse and may cause **high connection cost**!

Define the Used Language

The default language is defined by the telephony switch for various features

The language can be defined with parameter "Language" and applies to the following features:

- The AdminCenter Web GUI
- The standard announcements of the personal VoiceMail box
- The standard announcements of the telephony switch, for example, with an active "Do not Disturb"

On Busy

When the subscriber has an active connection and a further call is coming in then the new caller hears usually the busy tone.

As multiple VoIP devices may be registered to the telephone number the user can define the behavior of the feature "On Busy" :

◊ "Signal Busy":
Immediately rejects the new incoming call (no call waiting) when one device is busy. The caller hears the busy tone.
◊ "Start call waiting, busy if one device rejects":
All devices are signaled with the new incoming call. If one device is rejecting the new call then the caller hears the busy tone.
◊ "Start call-waiting, busy if all devices reject":
All devices are signaled with the new incoming call. If all devices are rejecting the new call then the caller hears the busy tone.

If there is just one device registered the later two options behave the same.

Assigning to a vPBX Department

If a vPBX is divided into departments then the telephone number can be assigned to a "Department" .

A department can be used:

◊ for sending SMS (the VoIP system must contain a SMS service) to all members of the department

Define the Location of the Telephone for Emergency Calls

If the subscriber places an emergency call, e.g. 112, the telephony switch routes the call to the nearest emergency call center.

In order to find the nearest emergency call center the telephony switch needs to know the current location of the subscriber. The current location has to be configured with parameter "Location (for Emergency)" . If no location is configured the telephony switch uses the default emergency call center.

Usage note for parameter "Location (for Emergency)" :
Since there may be several thousand locations available, a full selection list is unfavorable. Therefore the input field works as a search box. When letters and/or numbers are entered (e.g. of a community) then all locations are listed that match the entry. When the correct location is displayed it can be selected.

Limitation of the IP Subnet for the SIP Devices

With this feature, it is possible to define an IP subnet where the SIP devices are allowed to register at this user account. Registration attempts from SIP devices out of other IP subnets are rejected.

The IP subnet is defined with the parameter "Network" . It is possible to configure several IP subnets separated by commas.
Example:

```
172.1.1.0/24,192.168.10.0/24
```

Limitation of the IP Subnets with a SIP Profile for the SIP Devices

With this feature it is possible to define a set of IP subnets where the SIP devices are allowed to register at this user account. Registration attempts from SIP devices out of other IP subnets are rejected.

The provider prepares the available "SIP Profiles" . Check with the provider which IP subnets are associated with a SIP profile.

User account on the "UCC WebRTC Conference Service"

Note	This feature is only available if the "UCC WebRTC Conference Service" is set up on the VoIP system.

The user can define here whether a UCC user account on the "UCC WebRTC Conference Service" is activated. If the UCC user account is active , the user can initiate "UCC WebRTC Conferences" in addition to the normal audio conferences.

The username (email address) and password of the AdminCenter account are used for the login in the user account of the "UCC WebRTC Conference Service".

Call Recording

This feature allows the user to record his/hers conversations and to be sent as an audio file to a definable email address.

The feature is activated when the parameter "Recording E-Mail" is configured.

The recording of a conversation can be started as follows:

◊ The subscriber presses an appropriate key on his SIP telephone during the active call.
(Note: Not all SIP telephones support this feature with a configurable key.)

◊ The subscriber presses the following *# procedures on his telephone during the active call:
##1 : Start recording
##0 : Stop recording
(Note: DTMF must be activated on the telephone.)

◊ All calls of the subscriber are automatically recorded when parameter " Auto recording " is activated.

Configuration

Where to Configure this Feature

As vPBX administrator:

→ Tab "Subscriber xx"

→ Tab "Account"

As user:

→ Tab "Account"

Parameter Configuration

Parameter: Name

Description:	Defines a name to describe the telephone number. This information is transmitted within the vPBX as "Display Name" and can be displayed by a called SIP device.
Configuration:	Configuration String: ◊ Any string
Default:	None
Version:	AdminCenter V5.7

Parameter: Displayed external Number

Description:	Defines which public telephone number (CLIP) of the vPBX will be displayed at the called party in the public telephony network.
Configuration:	Selection Menu: None Selection Menu: Own public telephone number List of the public vPBX telephone numbers
Default:	Public vPBX telephone number with direct dialing in to the internal telephone number
Version:	AdminCenter V5.7

Parameter: Suppress own number

Description:	Defines which telephone number to be displayed at the called side: ◊ "Do not change" The information provided by the telephone or vPBX telephone number of the caller is transferred unchanged ◊ "Yes" The telephone number is always suppressed (CLIR) anonymous call ◊ "No" The public telephone number, which is used for the call is displayed (CLIP).
Configuration:	Selection Menu: Do not change Yes No
Default:	Do not change
Version:	AdminCenter V5.7

Parameter: Mobile Number

Description:	Defines the associated mobile telephone number of the subscriber. The mobile telephone number must be in international number notation, e.g.: `+411234567890`
Configuration:	
Default:	
Version:	AdminCenter V6.4

Parameter: Domain

Description:	SIP domain for the SIP device, which shall register to this telephone number.
Configuration:	Configuration String: ◊ IP address
Default:	
Version:	AdminCenter V5.7

Parameter: Authentication Name

Description:	SIP authenticating name for the SIP device, which shall register to this telephone number.
Configuration:	Configuration String: ◊ Any string **Note** Follow the online instructions concerning secure authenticating name!
Default:	
Version:	AdminCenter V5.7

Parameter: Password

Description:	SIP password for the SIP device, which shall register to this telephone number.
Configuration:	Configuration String: ◊ Any string **Note** Follow the online instructions concerning secure passwords!
Default:	
Version:	AdminCenter V5.7

Parameter: Language

Description:	Defines the language to be used in: Definiert, welche Sprache benutzt werden soll für: • The AdminCenter Web GUI • The standard announcements of personal voicemail box • The standard information of the telephone switch, e.g. an active "Do not Disturb"
Configuration:	Selection Menu: List of all available languages
Default:	
Version:	AdminCenter V5.7

Parameter: On Busy

Description:	Defines whether during an active call another incoming call is signaled by a warning tone and how the caller is signaled. ◊ "Signal Busy": Immediately rejects the new incoming call (no call waiting) when one device is busy. ◊ "Start call waiting, busy if one device rejects": All devices are signaled with the new incoming call. If one device is rejecting the new call then the caller hears the busy tone. ◊ "Start call-waiting, busy if all devices reject": All devices are signaled with the new incoming call. If all devices are rejecting the new call then the caller hears the busy tone.
Configuration:	Selection Menu: Signal Busy Start call waiting, busy if one device rejects Start call-waiting, busy if all devices reject
Default:	Start call waiting, busy if one device rejects
Version:	AdminCenter V6.4

Parameter: Location (for Emergency)

Description:	Defines the location of the telephone for handling emergency calls, e.g. 112 **Note** It is important that the location is known as precisely as possible. In an emergency the telephony switch can route the call to the correct emergency center.
Configuration:	Configuration String: ◊ List of all defined locations Usage note: The input works as a search box. When letters and/or numbers are entered (e.g. a community) then all locations are listed that match the entry. When the correct location is displayed then it can be selected.
Default:	None
Version:	AdminCenter V5.7

Parameter: Recording E-Mail

Description:	Defines to which email address the attached audio file is sent with the recorded conversation. If no email address is entered, the feature is not activated!
Configuration:	Configuration String: ◊ None : No call recording active ◊ Email Address
Default:	None
Version:	AdminCenter V5.7

Parameter: Auto recording

Description:	Defines whether all conversations of the subscriber will be automatically recorded. The individual calls are sent as an audio file to the specified email address.
Configuration:	Selection Button: ☑ Activated - ☐ Not activated
Default:	Not activated
Version:	AdminCenter V5.8

Parameter: Network

Description:	Defines from which IP subnet a SIP device is allowed to register at this user account. The network is defined by the starting IP address and its subnet mask. Several comma separated IP subnets are possible.

	Example: `172.1.1.0/24,192.168.10.0/24`
Configuration:	Configuration String: ◊ None : No IP address checking ◊ IP address / Subnet Mask Bit
Default:	None
Version:	AdminCenter V6.2

Parameter: SIP Profile

Description:	Defines a SIP Profile which contains a list of IP subnets where a SIP device is allowed to register at this user account. Check with the provider which IP subnets are associated with a SIP profile.
Configuration:	Selection Menu: - (None) List of prepared SIP Profiles
Default:	None
Version:	AdminCenter V6.2

Parameter: Department

Description:	Defines a vPBX department for this telephone number.
Configuration:	Selection Menu: - (None) List of prepared departments
Default:	None
Version:	AdminCenter V6.4

Parameter: UCC User

Description:	Defines that the user has a UCC user account on the "UCC WebRTC Conference Service" . The UCC account is generated automatically on the "UCC WebRTC Conferencing" service. Afterwards the user can initiate "UCC WebRTC Conferences".
Configuration:	Selection Button: ☑ Activated - ☐ Not activated
Default:	Not activated
Version:	AdminCenter V6.8