EdgeMarc 2900 POE Interop with Swyx PBX - Use Case 2

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Interoperable Vendors

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Document Overview

This document outlines the configuration best practices for Vodafone involving EdgeMarc 2900 when deployed with SwyxPBX. This document provides the configuration snapshot of the interoperability performed between Ribbon's EdgeMarc 2900 and SwyxPBX, SwyxIt and Swyx Phone. SwyxPBX is a fully **"Hosted PBX Service**" with cloud telephony. In cases when you no longer have a PBX in your company, you obtain all features as a service from the cloud but you can keep your existing contract with your Service Provider who will provide you a local breakout.

Scope

This document provides configuration best practices for deploying Ribbon's EdgeMarc 2900 with SwyxPBX and associated users. Note that these are configuration best practices and each customer may have unique needs and networks. Ribbon recommends that customers work with network design and deployment engineers to establish the network design which best meets their requirements.

Non-Goals

It is not the goal of this document to provide detailed configurations that will meet the requirements of every customer. Use this document as a starting point and build the SBC configurations in consultation with network design and deployment engineers.

Audience

This is a technical document is intended for telecommunications engineers with the purpose of configuring both the Ribbon EdgeMarc 2900 and the SwyxPBX and associated users.

Steps will require navigating the third-party product as well as the Ribbon product using graphical user interface (GUI) or command line interface (CLI). An understanding of the basic concepts of TCP/UDP/TLS, IP/Routing, and SIP/RTP/SRTP is needed to complete the configuration and any necessary troubleshooting.

Pre-Requisites

The following aspects would be required before proceeding with Ribbon EM 2900 POE & SwyxWare 12.10.

- SwyxWare 12.10 is installed in a Windows Server Platform and connected to the network.
- A 190 trial license is available and obtained from Swyx.
- Remote Desktop access to Windows host is available for remote access and configuration.
- A valid 6 months trial License is running on the Server.
- HFA firmware is loaded and installed on to Unify CP205 Phone Unit.

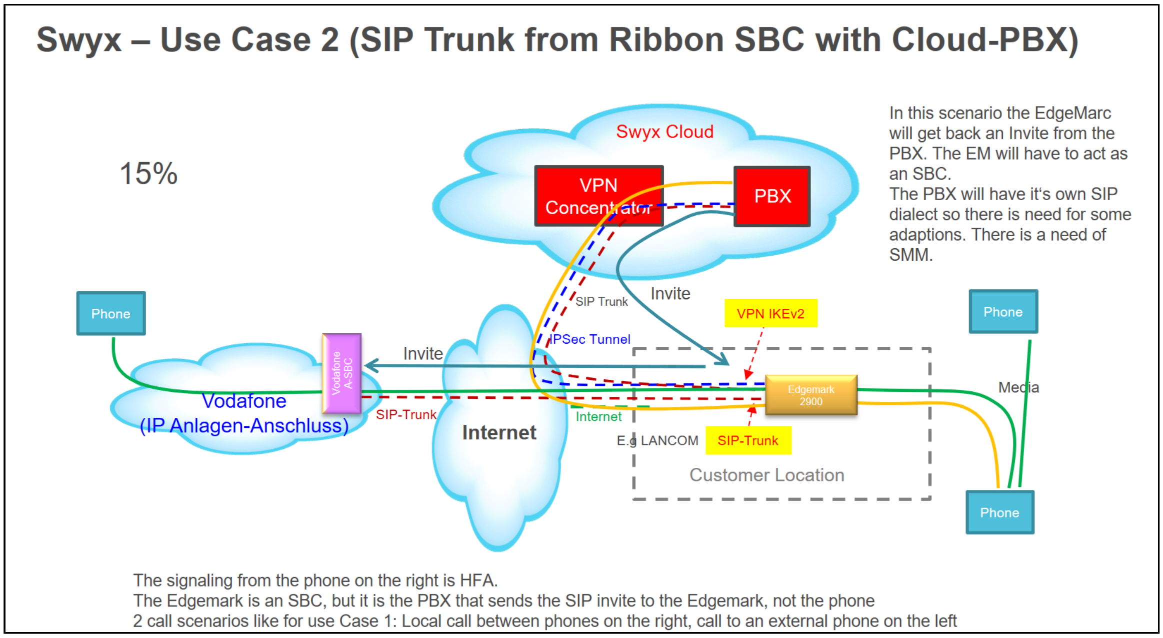
Product and Device Details

	Equipment/Product	**Software Version**
Ribbon Communications	EdgeMarc 2900 POE	Version 15.8.3
Third-Party Products	SwyxWare	V12.10.16296.0
	SwyxIt	V12.10.16296
	Windows Server	2019
	Unify CP205	V1 R3.9.0 HFA 190516

Network Topology Diagram

Use Case 2 Deployment Topology

The deployment topology diagram is depicted below.



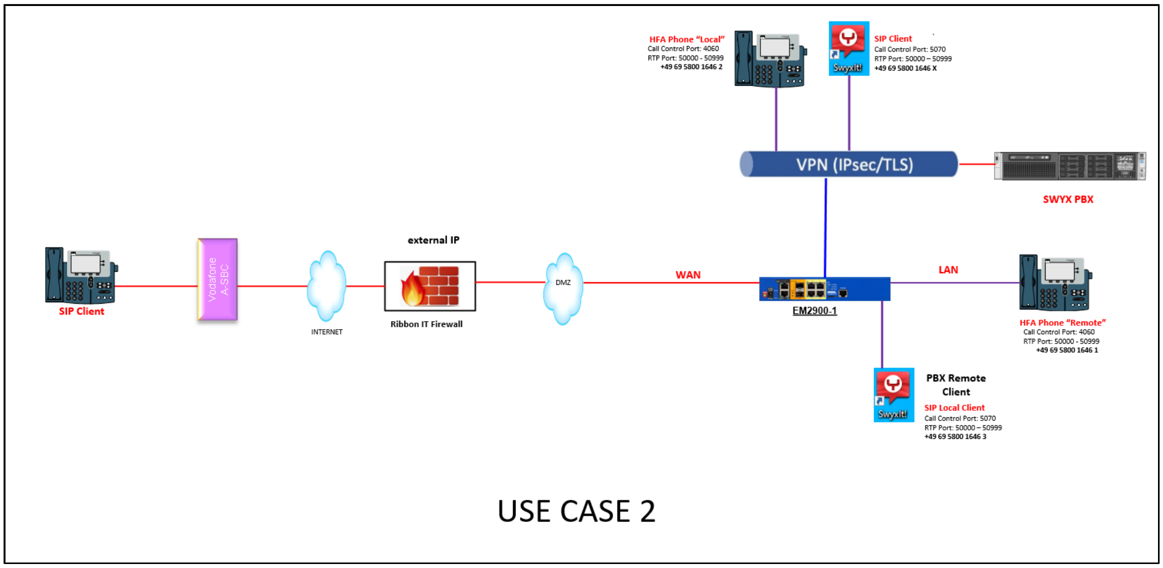
Swyx – Use Case 2 (SIP Trunk from Ribbon SBC with Cloud-PBX)

In this scenario the EdgeMarc will get back an Invite from the PBX. The EM will have to act as an SBC.
The PBX will have it's own SIP dialect so there is need for some adaptions. There is a need of SMM.

The signaling from the phone on the right is HFA.
The Edgemark is an SBC, but it is the PBX that sends the SIP invite to the Edgemark, not the phone
2 call scenarios like for use Case 1: Local call between phones on the right, call to an external phone on the left

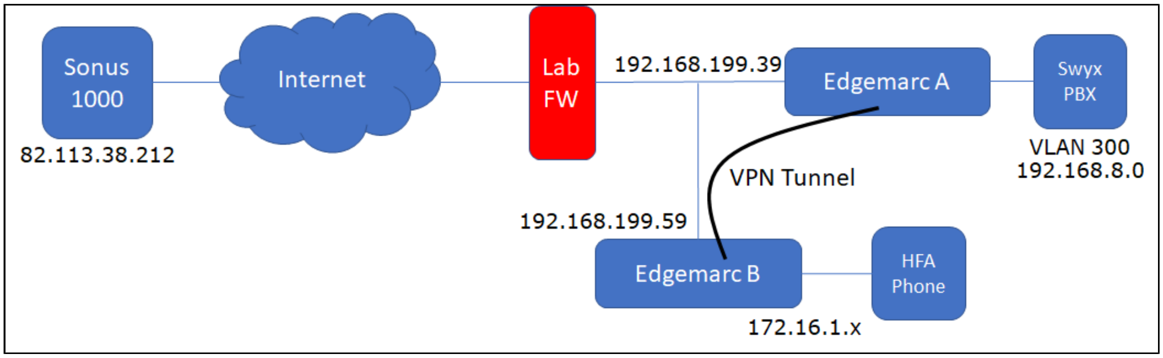
Interoperability Test Lab Topology (or Call Flow Diagram)

IOT high-level architecture covering call flows & overall topology is depicted below.



Section-A: EdgeMarc Configuration

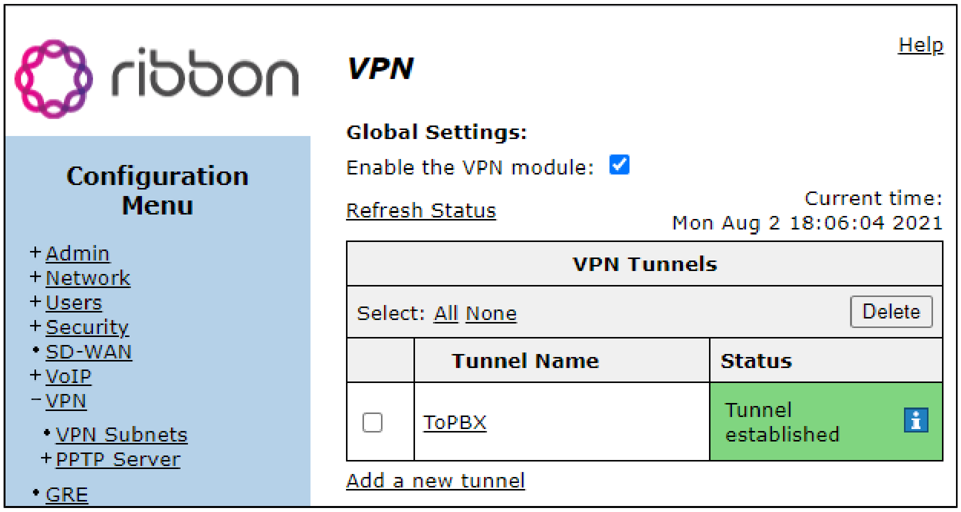
Connectivity



Configuring EdgeMarc A

In this use case, EdgeMarc A has a simple VPN configuration with no SIP configuration.

1. Enable the VPN module then select Add a new VPN tunnel.



VPN

Global Settings:

Enable the VPN module: ☑

Current time:
Mon Aug 2 18:06:04 2021

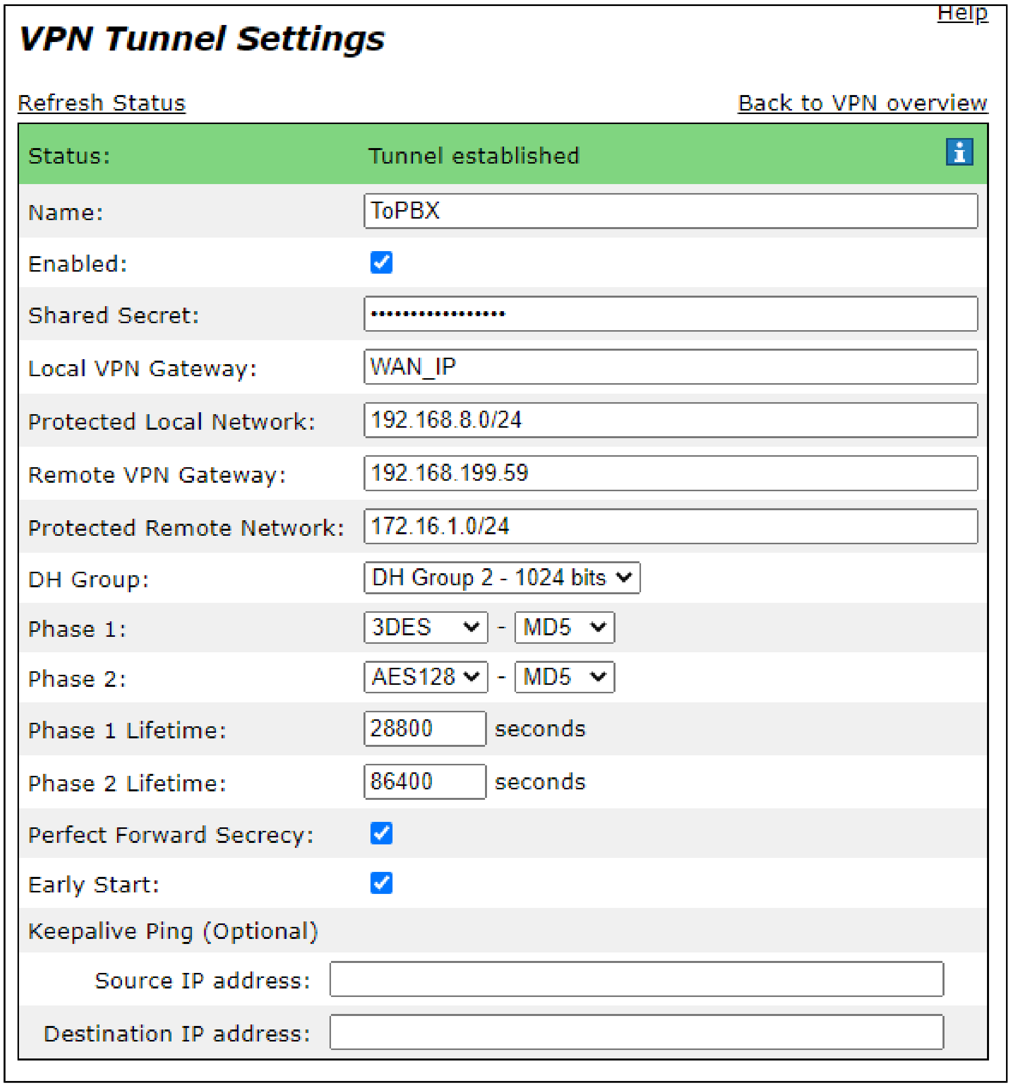
VPN Tunnels

Select: All None

	Tunnel Name	Status
□	ToPBX	Tunnel established

2. Configure the following:
   - The Protected Local Network - voice VLAN of EdgeMarc A.
   - The Remote VPN Gateway - WAN IP of EdgeMarc B.

- The Protected Remote network - voice VLAN of EdgeMarc B.



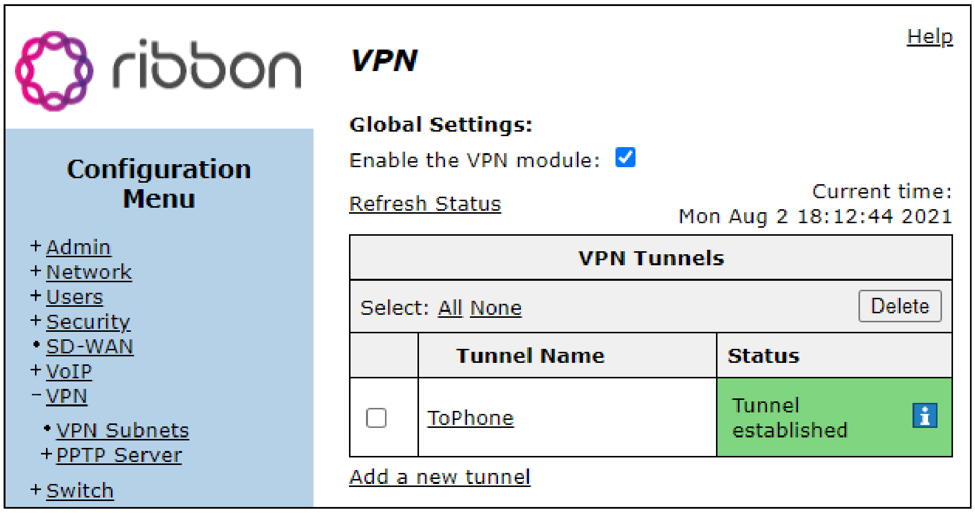
VPN Tunnel Settings

Status:	Tunnel established
Name:	ToPBX
Enabled:	☑
Shared Secret:	•••••••••••••••••
Local VPN Gateway:	WAN_IP
Protected Local Network:	192.168.8.0/24
Remote VPN Gateway:	192.168.199.59
Protected Remote Network:	172.16.1.0/24
DH Group:	DH Group 2 - 1024 bits
Phase 1:	3DES - MD5
Phase 2:	AES128 - MD5
Phase 1 Lifetime:	28800 seconds
Phase 2 Lifetime:	86400 seconds
Perfect Forward Secrecy:	☑
Early Start:	☑
Keepalive Ping (Optional)	
Source IP address:	
Destination IP address:	

Configuring EdgeMarc B

EdgeMarc B is responsible for communicating with the SIP Service Provider as well as passing HFA phone traffic via IPSec tunnel to the head-end PBX. Configure IPSec:

1. Enable the VPN module then select Add a new tunnel.



VPN

Global Settings:

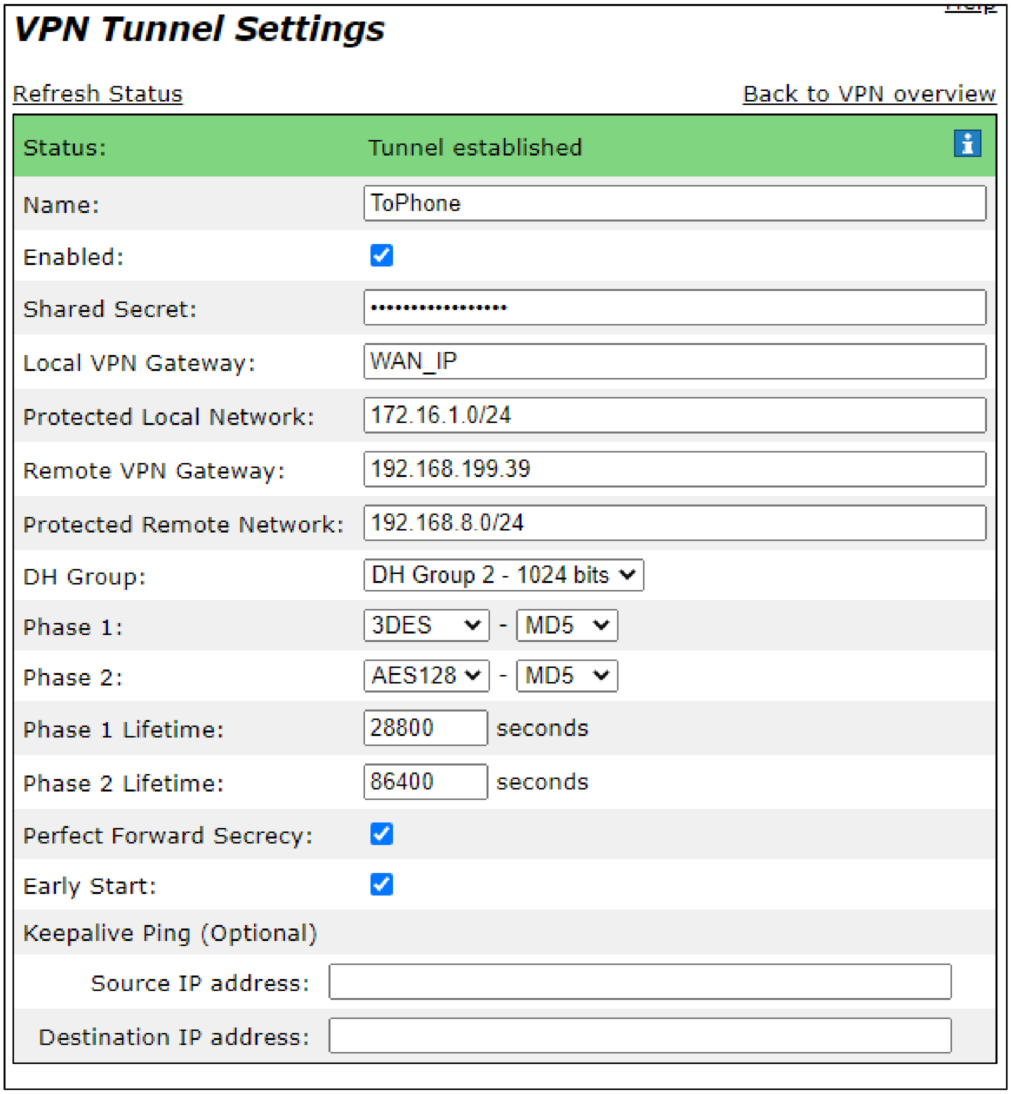
Enable the VPN module: ☑

Current time: Mon Aug 2 18:12:44 2021

VPN Tunnels

	Tunnel Name	Status
☐	ToPhone	Tunnel established

2. Configure the following:
    - The Protected Local Network - voice VLAN of EdgeMarc B.
    - The Remote VPN Gateway - WAN IP of EdgeMarc A
    - The Protected Remote network - voice VLAN of EdgeMarc A.



VPN Tunnel Settings

Help

Refresh Status | Back to VPN overview

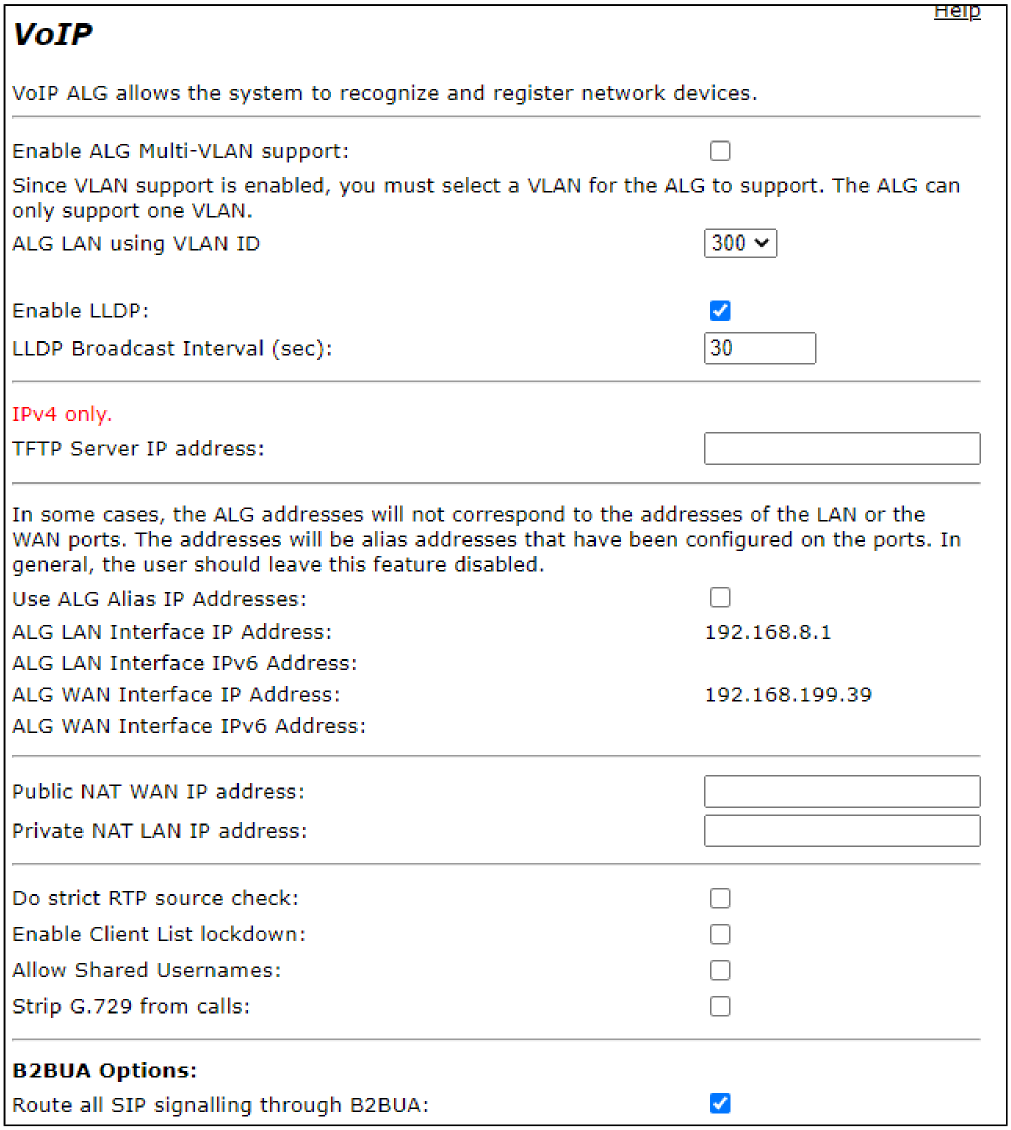
Status:	Tunnel established
Name:	ToPhone
Enabled:	☑
Shared Secret:	••••••••••••••••••
Local VPN Gateway:	WAN_IP
Protected Local Network:	172.16.1.0/24
Remote VPN Gateway:	192.168.199.39
Protected Remote Network:	192.168.8.0/24
DH Group:	DH Group 2 - 1024 bits
Phase 1:	3DES - MD5
Phase 2:	AES128 - MD5
Phase 1 Lifetime:	28800 seconds
Phase 2 Lifetime:	86400 seconds
Perfect Forward Secrecy:	☑
Early Start:	☑
Keepalive Ping (Optional)	
Source IP address:	
Destination IP address:	

In addition, a static route is required on EdgeMarc B to allow the ALG to function correctly when communicating to a SIP PBX across a VPN tunnel. After logging into the EdgeMarc, execute the following commands:

static router

```
echo "router add -net 192.168.8.0/24 gw 172.16.1.1" >> /etc/config/user_defs.conf
cfg_commit
config_network
```

3. Next, configure the ALG and SIP settings starting with the main VoIP page.



Help

VoIP

VoIP ALG allows the system to recognize and register network devices.

Enable ALG Multi-VLAN support: □

Since VLAN support is enabled, you must select a VLAN for the ALG to support. The ALG can only support one VLAN.

ALG LAN using VLAN ID: 300

Enable LLDP: ☑

LLDP Broadcast Interval (sec): 30

IPv4 only.

TFTP Server IP address:

In some cases, the ALG addresses will not correspond to the addresses of the LAN or the WAN ports. The addresses will be alias addresses that have been configured on the ports. In general, the user should leave this feature disabled.

Use ALG Alias IP Addresses: □

ALG LAN Interface IP Address: 192.168.8.1

ALG LAN Interface IPv6 Address:

ALG WAN Interface IP Address: 192.168.199.39

ALG WAN Interface IPv6 Address:

Public NAT WAN IP address:

Private NAT LAN IP address:

Do strict RTP source check: □

Enable Client List lockdown: □

Allow Shared Usernames: □

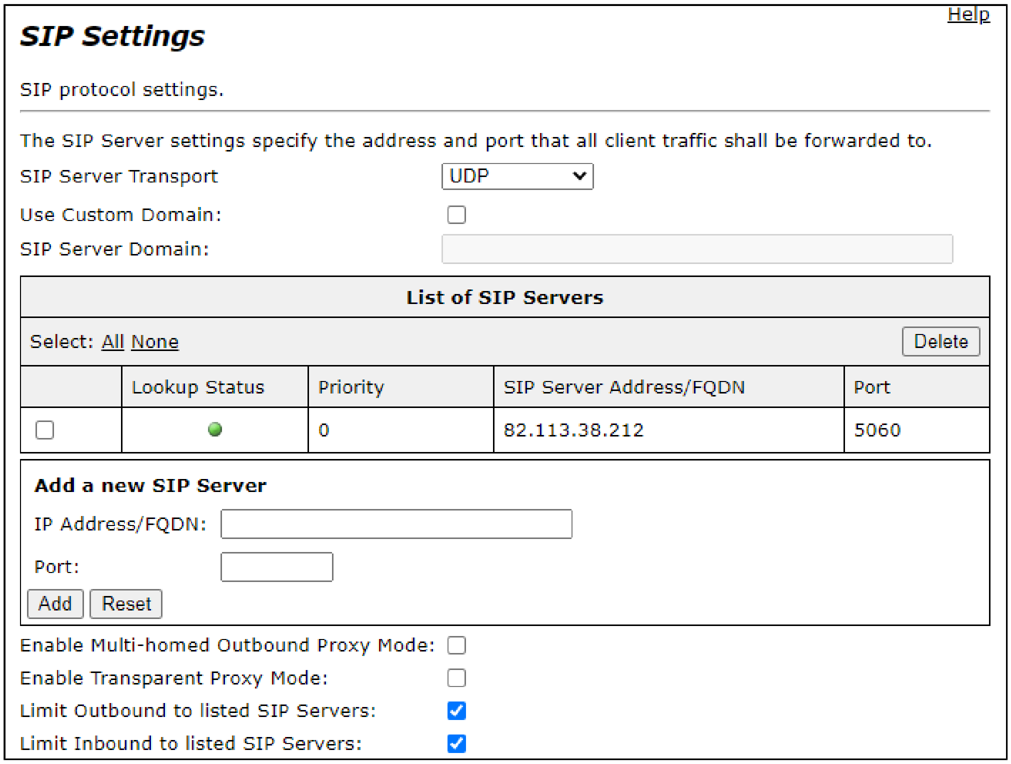
Strip G.729 from calls: □

B2BUA Options:

Route all SIP signalling through B2BUA: ☑

The ALG VLAN selected should correspond to the voice VLAN of the EdgeMarc.

4. Configure the SBC IP of the SIP Provider under list of SIP Servers. In this case the IP corresponds to the public IP of the Sonus 1000 used for testing.



SIP Settings

SIP protocol settings.

The SIP Server settings specify the address and port that all client traffic shall be forwarded to.

SIP Server Transport	UDP
Use Custom Domain:	□
SIP Server Domain:	

List of SIP Servers

Select: All None

	Lookup Status	Priority	SIP Server Address/FQDN	Port
□	●	0	82.113.38.212	5060

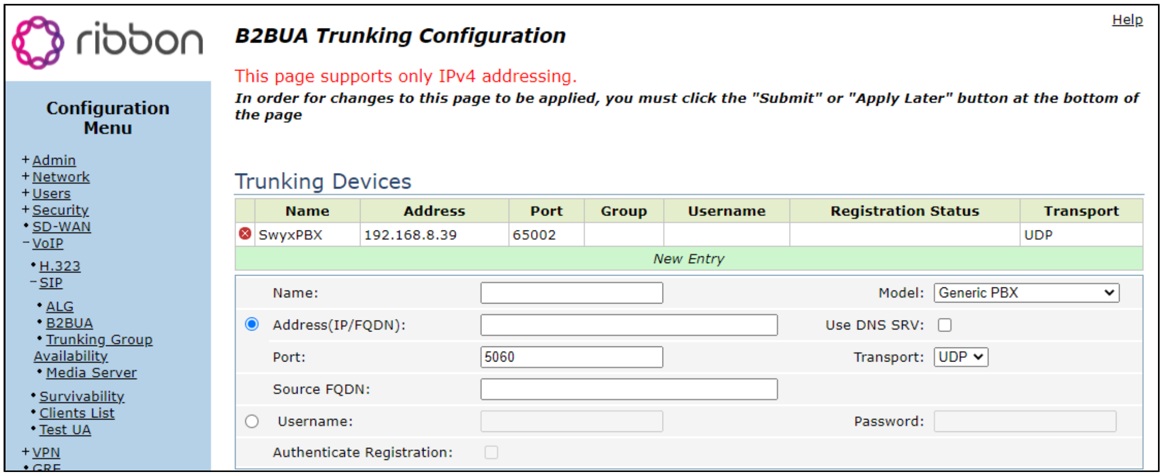
Add a new SIP Server

IP Address/FQDN:

Port:

Enable Multi-homed Outbound Proxy Mode:	□
Enable Transparent Proxy Mode:	□
Limit Outbound to listed SIP Servers:	☑
Limit Inbound to listed SIP Servers:	☑

5. Next configure the LAN side SIP services.



B2BUA Trunking Configuration

This page supports only IPv4 addressing.

In order for changes to this page to be applied, you must click the "Submit" or "Apply Later" button at the bottom of the page

Trunking Devices

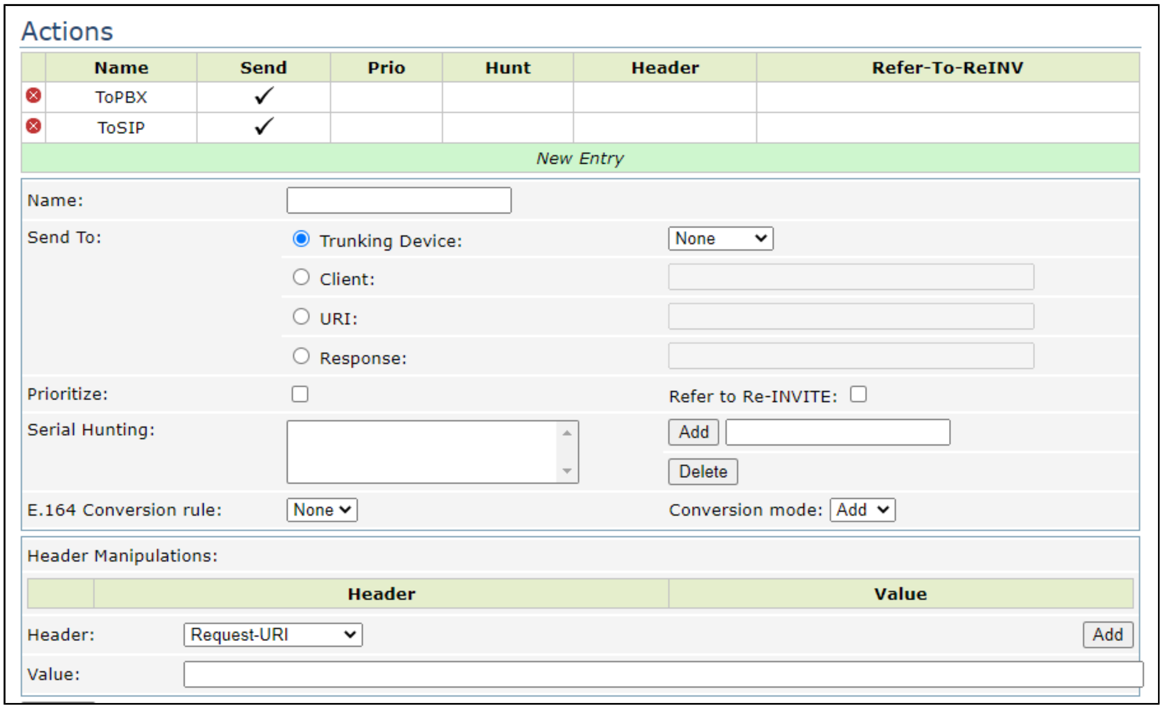
	Name	Address	Port	Group	Username	Registration Status	Transport
☒	SwyxPBX	192.168.8.39	65002				UDP

New Entry

	Name:		Model:	Generic PBX
●	Address(IP/FQDN):		Use DNS SRV:	□
	Port:	5060	Transport:	UDP
	Source FQDN:			
○	Username:		Password:	
	Authenticate Registration:	□		

A trunking device is configured so that the EdgeMarc knows where to forward an inbound call from the SIP service provider. In this case, a non-standard port is used due to requirements of the Swyx PBX.

6. Actions are configured in order to facilitate call routing in the Match section.



Actions

	Name	Send	Prio	Hunt	Header	Refer-To-ReINV
☒	ToPBX	✓				
☒	ToSIP	✓				

New Entry

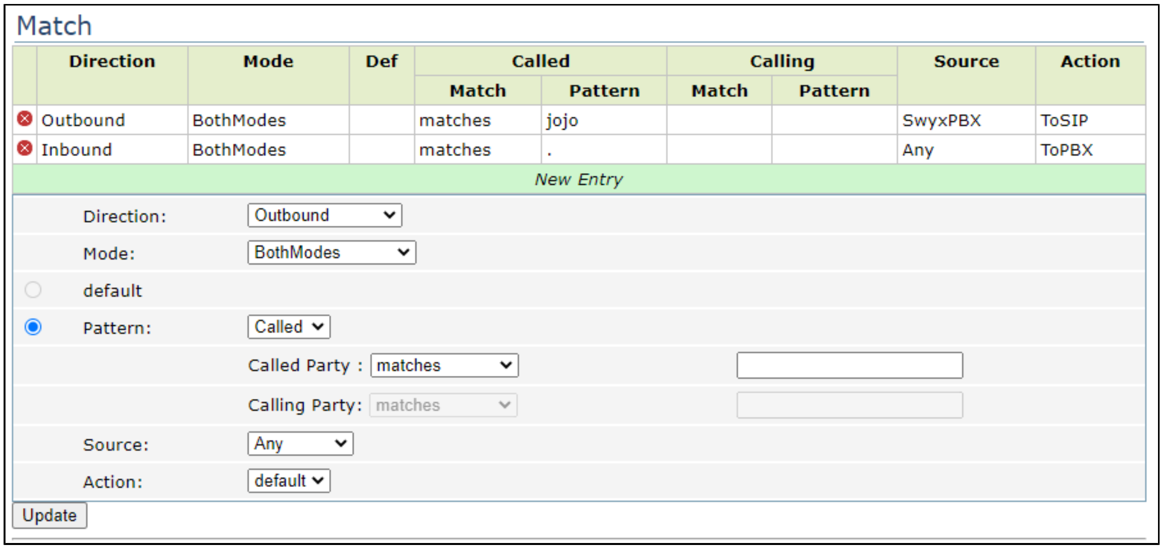
Name:			
Send To:	● Trunking Device:	None	
	○ Client:		
	○ URI:		
	○ Response:		
Prioritize:	□	Refer to Re-INVITE: □	
Serial Hunting:		Add	
		Delete	
E.164 Conversion rule:	None	Conversion mode:	Add

Header Manipulations:

	Header	Value

Header: Request-URI Add

Value:



Match

	Direction	Mode	Def	Called		Calling		Source	Action
				Match	Pattern	Match	Pattern		
☒	Outbound	BothModes		matches	jojo			SwyxPBX	ToSIP
☒	Inbound	BothModes		matches	.			Any	ToPBX

New Entry

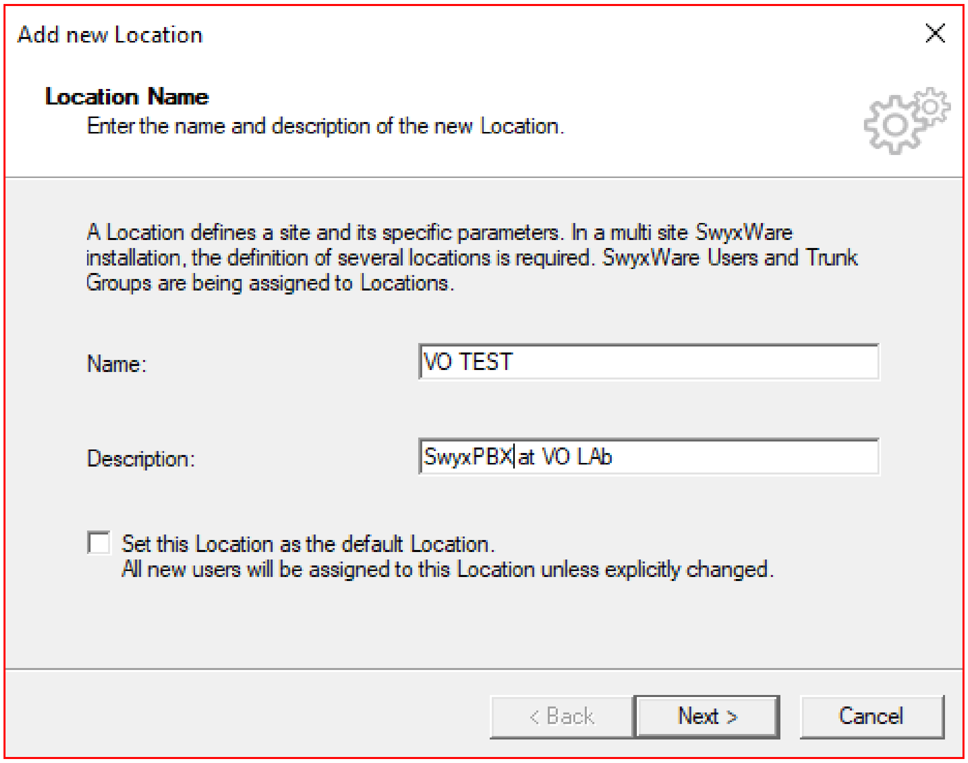
	Direction:	Outbound		
	Mode:	BothModes		
○	default			
●	Pattern:	Called		
		Called Party :	matches	
		Calling Party:	matches	
	Source:	Any		
	Action:	default		

The lab uses basic call routing that routes all inbound calls to the PBX and a specific client towards the provider.

Section-B: SwyxWare, SwyxIt and HFA Phone Configuration

Configuring SwyxPBX

1. Right Click on Location > Add Location.



Add new Location

Location Name

Enter the name and description of the new Location.

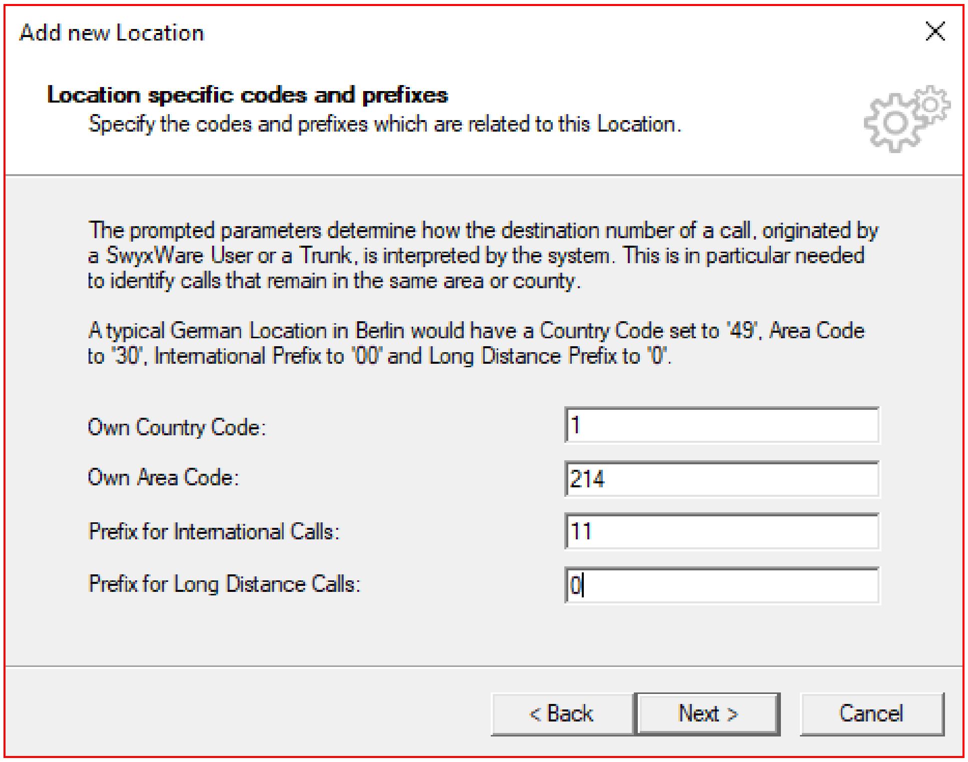
A Location defines a site and its specific parameters. In a multi site SwyxWare installation, the definition of several locations is required. SwyxWare Users and Trunk Groups are being assigned to Locations.

Name: VO TEST

Description: SwyxPBX at VO LAb

□ Set this Location as the default Location.
All new users will be assigned to this Location unless explicitly changed.

2. Add codes and prefixes then click Next.



Add new Location

Location specific codes and prefixes

Specify the codes and prefixes which are related to this Location.

The prompted parameters determine how the destination number of a call, originated by a SwyxWare User or a Trunk, is interpreted by the system. This is in particular needed to identify calls that remain in the same area or county.

A typical German Location in Berlin would have a Country Code set to '49', Area Code to '30', International Prefix to '00' and Long Distance Prefix to '0'.

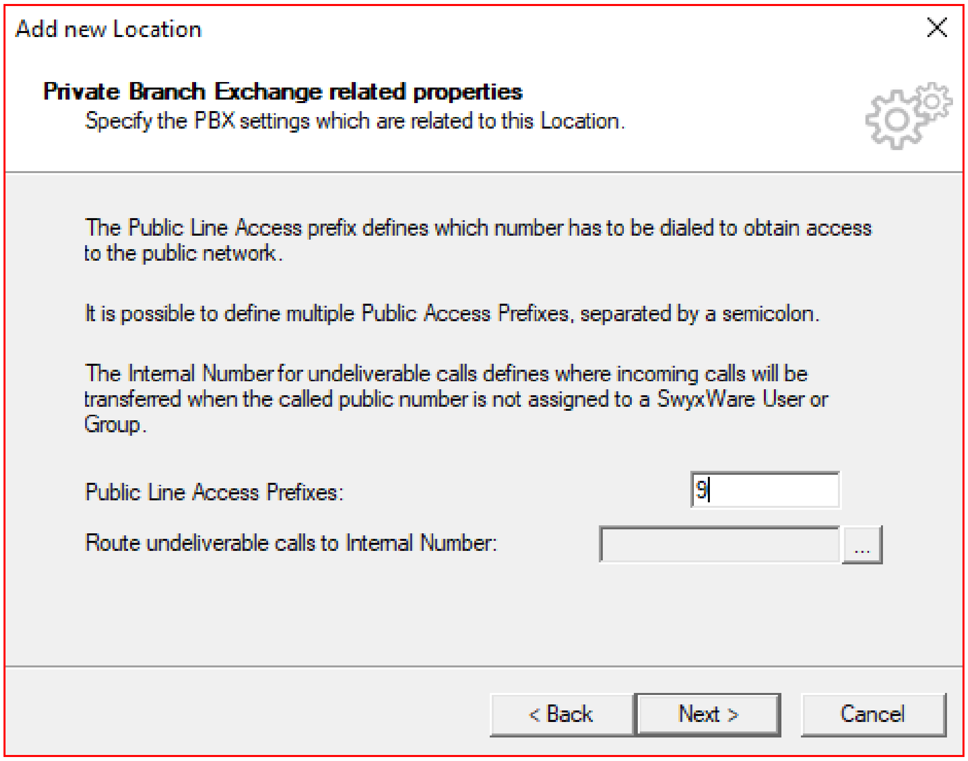
Own Country Code: 1

Own Area Code: 214

Prefix for International Calls: 11

Prefix for Long Distance Calls: 0

3. Add access to Dial out and click Next.



Add new Location

Private Branch Exchange related properties

Specify the PBX settings which are related to this Location.

The Public Line Access prefix defines which number has to be dialed to obtain access to the public network.

It is possible to define multiple Public Access Prefixes, separated by a semicolon.

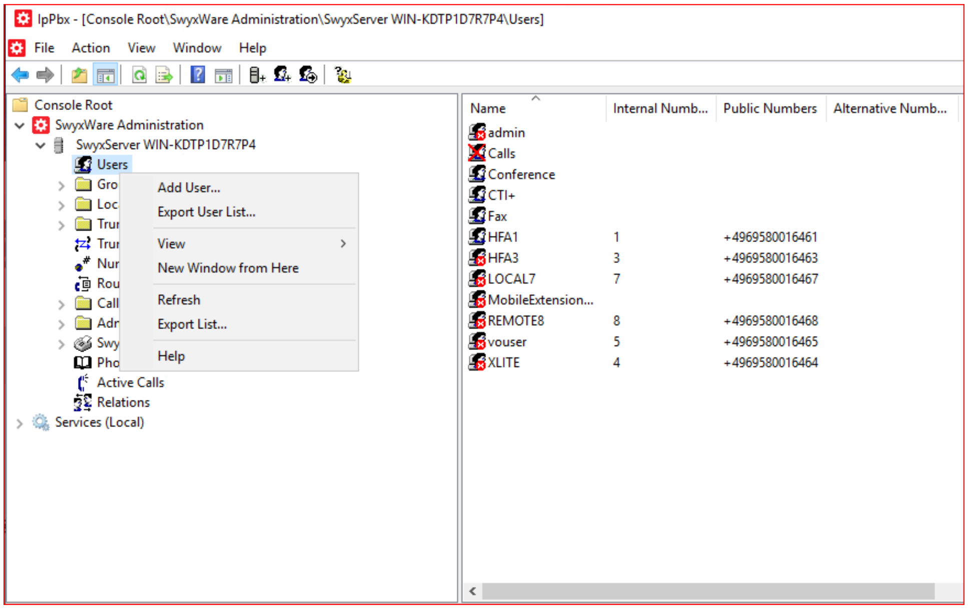
The Internal Number for undeliverable calls defines where incoming calls will be transferred when the called public number is not assigned to a SwyxWare User or Group.

Public Line Access Prefixes: 9

Route undeliverable calls to Internal Number:

Adding a User

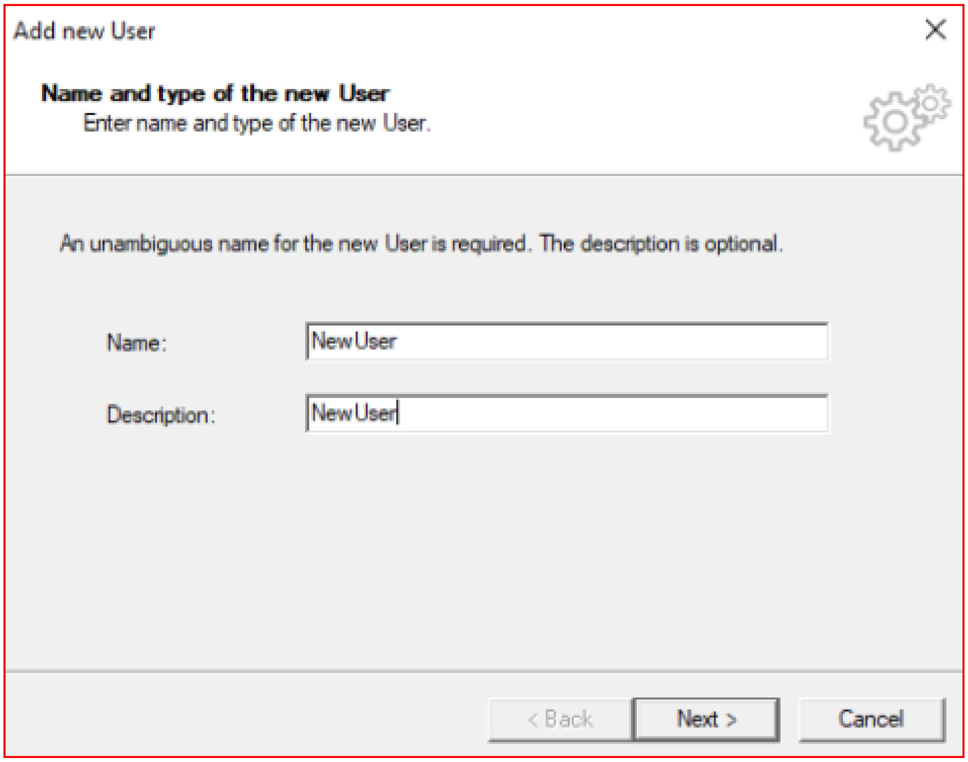
1. Click on Server > Right Click on User > click Add User.



IpPbx - [Console Root\SwyxWare Administration\SwyxServer WIN-KDTP1D7R7P4\Users]

Name	Internal Numb...	Public Numbers	Alternative Numb...
admin			
Calls			
Conference			
CTI+			
Fax			
HFA1	1	+4969580016461	
HFA3	3	+4969580016463	
LOCAL7	7	+4969580016467	
MobileExtension...			
REMOTE8	8	+4969580016468	
vouser	5	+4969580016465	
XLITE	4	+4969580016464	

2. Add Name and Description then Click Next.



Add new User

Name and type of the new User

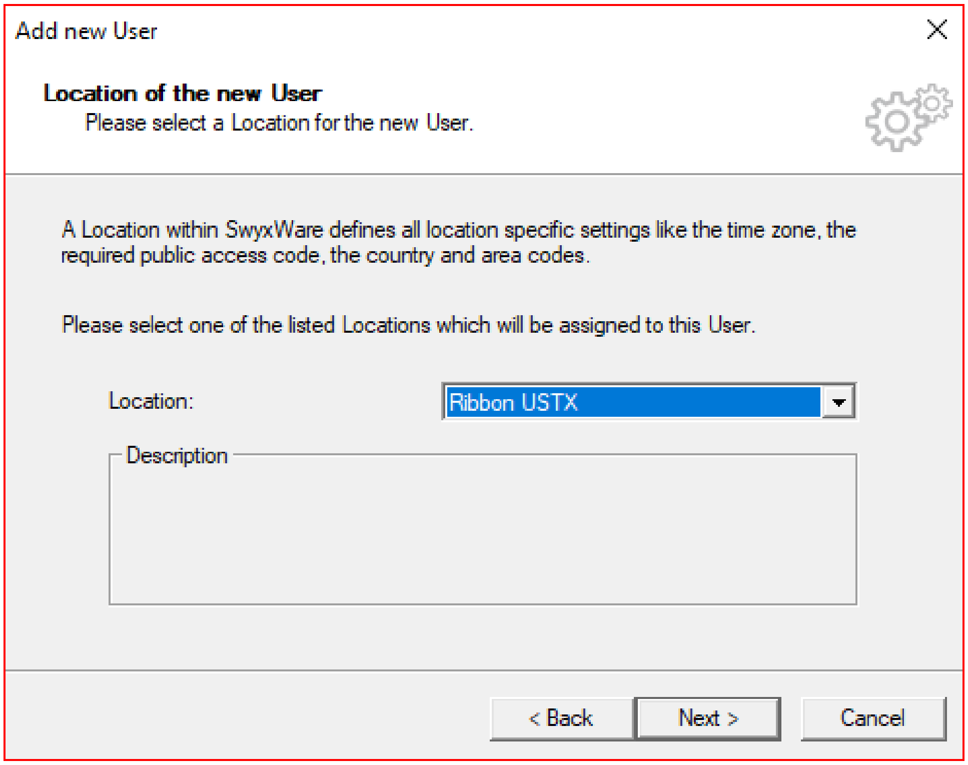
Enter name and type of the new User.

An unambiguous name for the new User is required. The description is optional.

Name: New User

Description: New User

3. Select the Location and click Next.



Add new User

Location of the new User

Please select a Location for the new User.

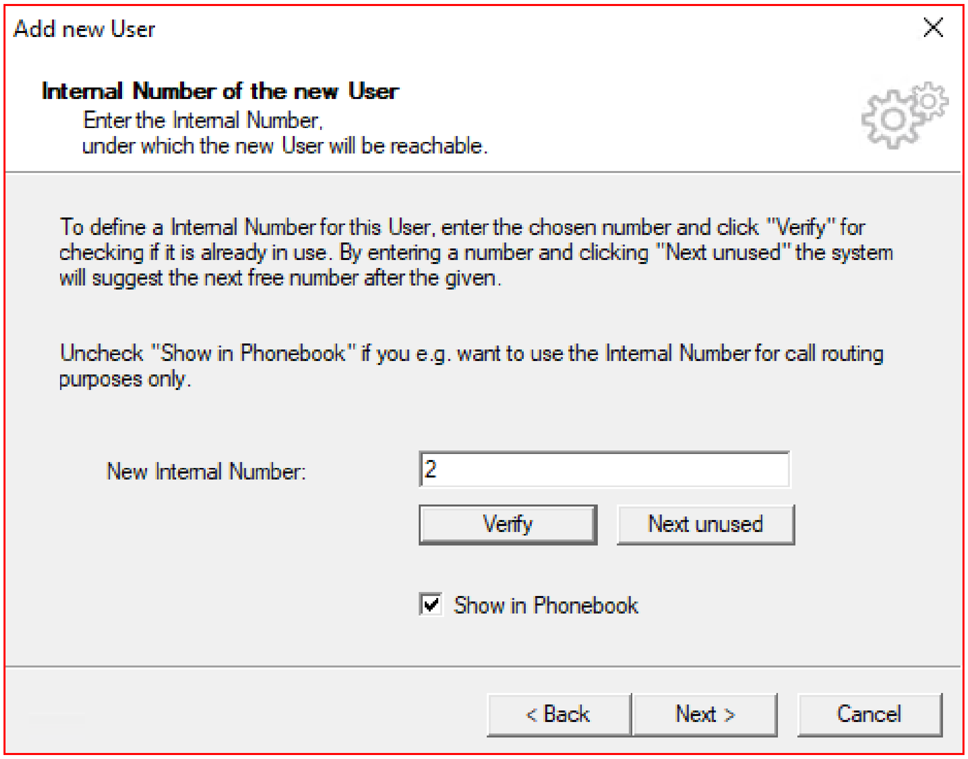
A Location within SwyxWare defines all location specific settings like the time zone, the required public access code, the country and area codes.

Please select one of the listed Locations which will be assigned to this User.

Location: Ribbon USTX

Description

4. Select a new internal number and click verify to check if available. Click OK then click next.



Add new User

Internal Number of the new User

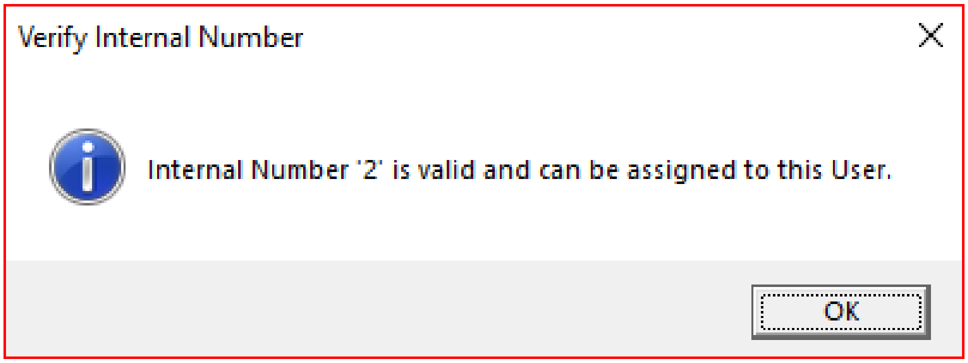
Enter the Internal Number,
under which the new User will be reachable.

To define a Internal Number for this User, enter the chosen number and click "Verify" for checking if it is already in use. By entering a number and clicking "Next unused" the system will suggest the next free number after the given.

Uncheck "Show in Phonebook" if you e.g. want to use the Internal Number for call routing purposes only.

New Internal Number: 2

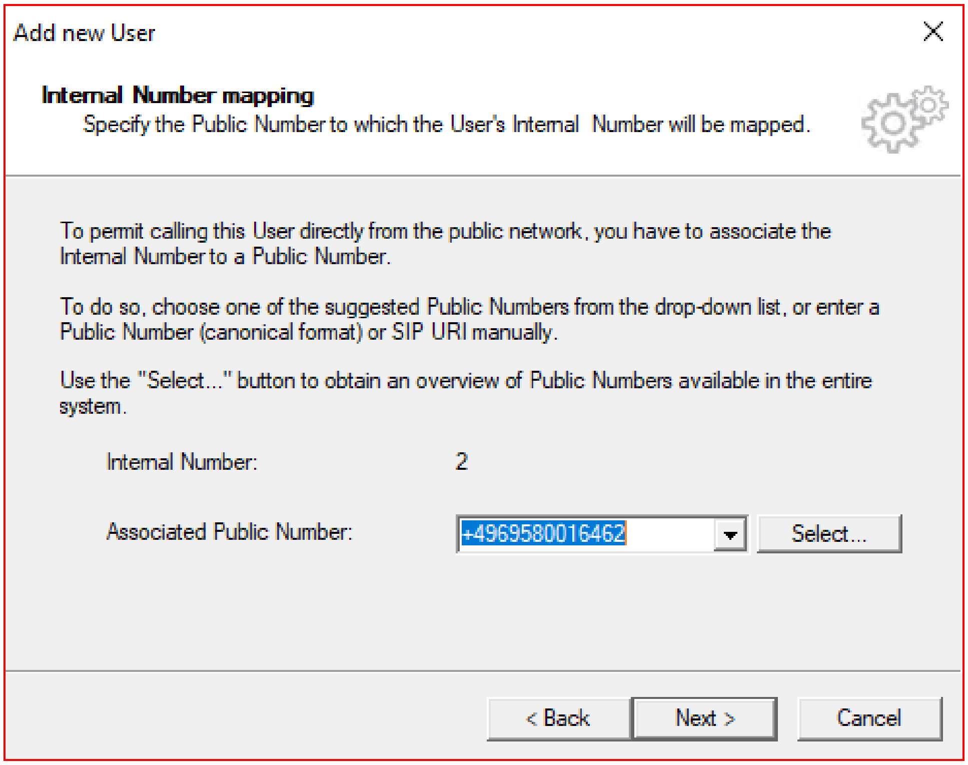
☑ Show in Phonebook



Verify Internal Number

Internal Number '2' is valid and can be assigned to this User.

5. The Internal Number selected will be mapped to a public number, then click Next.



Internal Number mapping

Specify the Public Number to which the User's Internal Number will be mapped.

To permit calling this User directly from the public network, you have to associate the Internal Number to a Public Number.

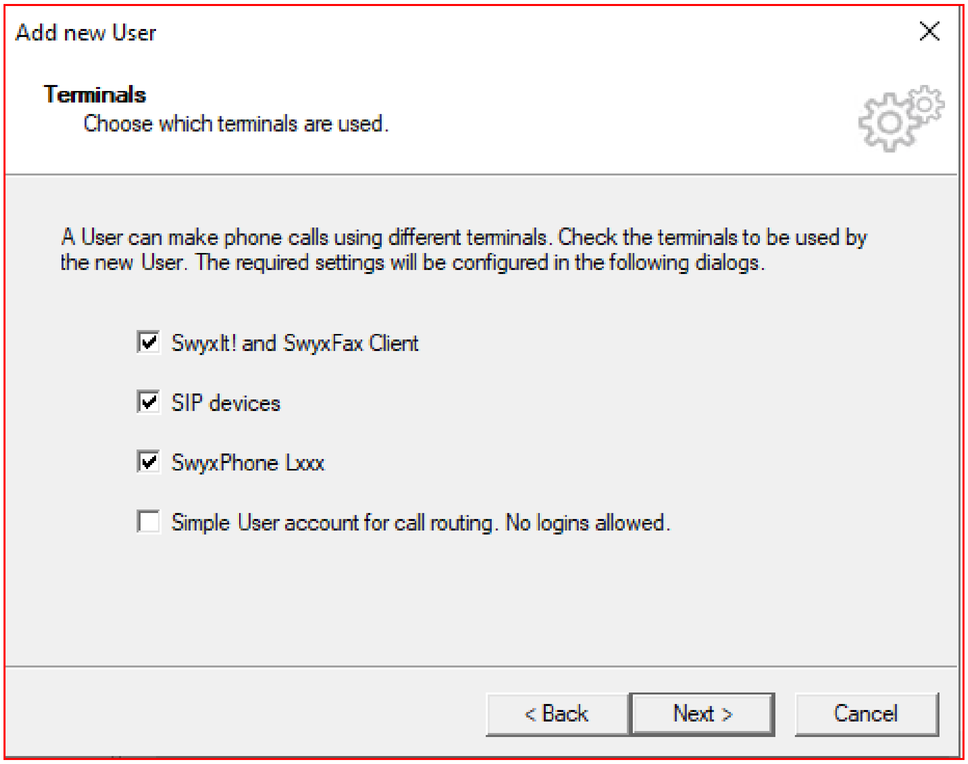
To do so, choose one of the suggested Public Numbers from the drop-down list, or enter a Public Number (canonical format) or SIP URI manually.

Use the "Select..." button to obtain an overview of Public Numbers available in the entire system.

Internal Number: 2

Associated Public Number: +4969580016462

6. Select the Terminals by checking boxes and then click Next.



Terminals

Choose which terminals are used.

A User can make phone calls using different terminals. Check the terminals to be used by the new User. The required settings will be configured in the following dialogs.

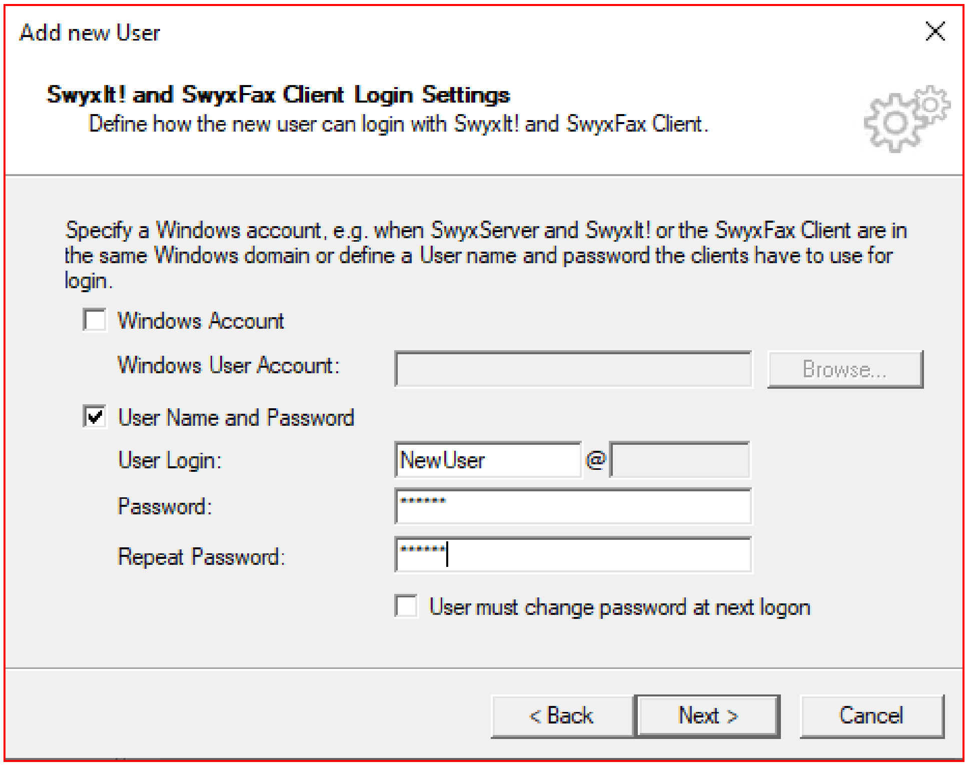
☑ SwyxIt! and SwyxFax Client

☑ SIP devices

☑ SwyxPhone Lxxx

☐ Simple User account for call routing. No logins allowed.

7. Create a Password for the user login then click Next.



Add new User

SwyxIt! and SwyxFax Client Login Settings

Define how the new user can login with SwyxIt! and SwyxFax Client.

Specify a Windows account, e.g. when SwyxServer and SwyxIt! or the SwyxFax Client are in the same Windows domain or define a User name and password the clients have to use for login.

□ Windows Account

Windows User Account:

☑ User Name and Password

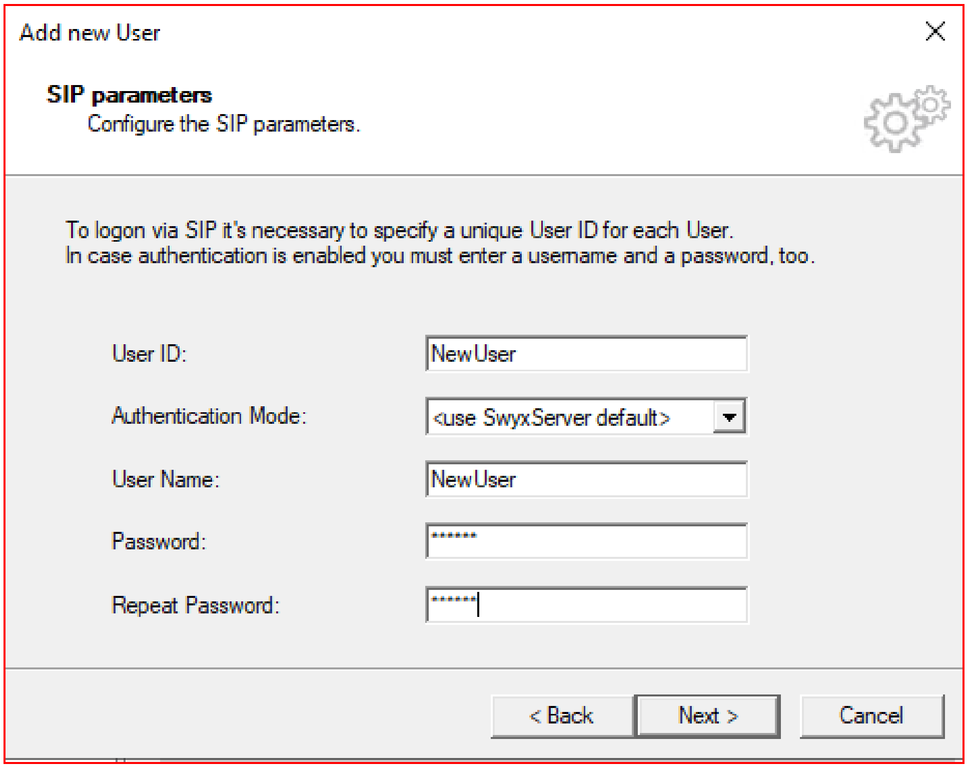
User Login: NewUser @

Password: ******

Repeat Password: ******

□ User must change password at next logon

8. Create SIP user and password and then click Next.



Add new User

SIP parameters

Configure the SIP parameters.

To logon via SIP it's necessary to specify a unique User ID for each User.
In case authentication is enabled you must enter a username and a password, too.

User ID: NewUser

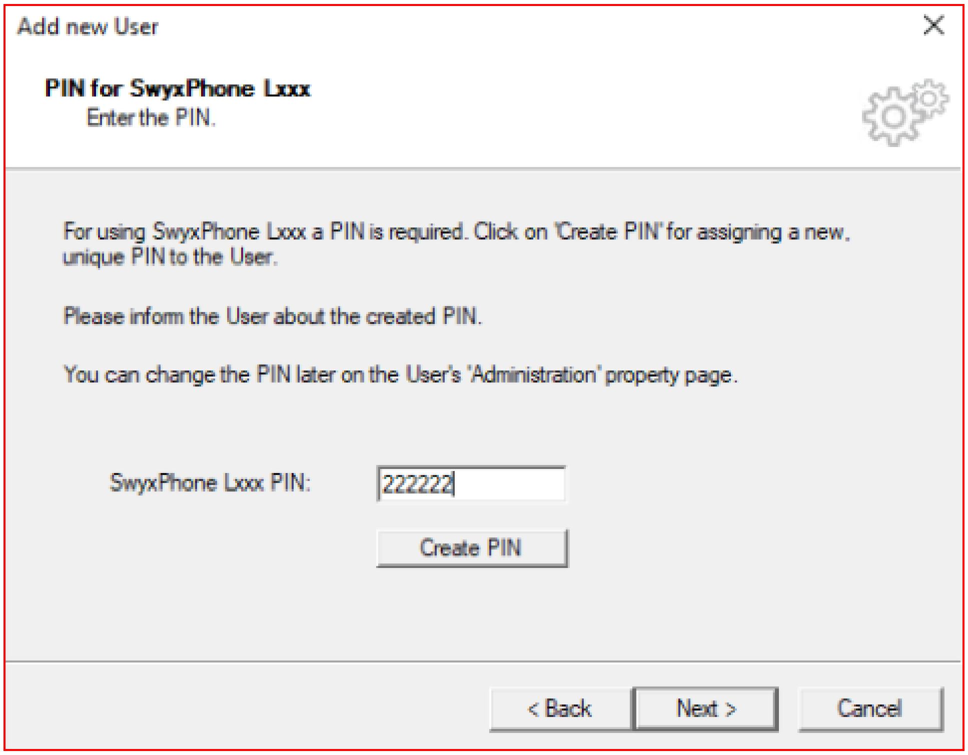
Authentication Mode: <use SwyxServer default>

User Name: NewUser

Password: ******

Repeat Password: ******

9. Create a Swyx Phone Pin and then click Next.



Add new User

PIN for SwyxPhone Lxxx

Enter the PIN.

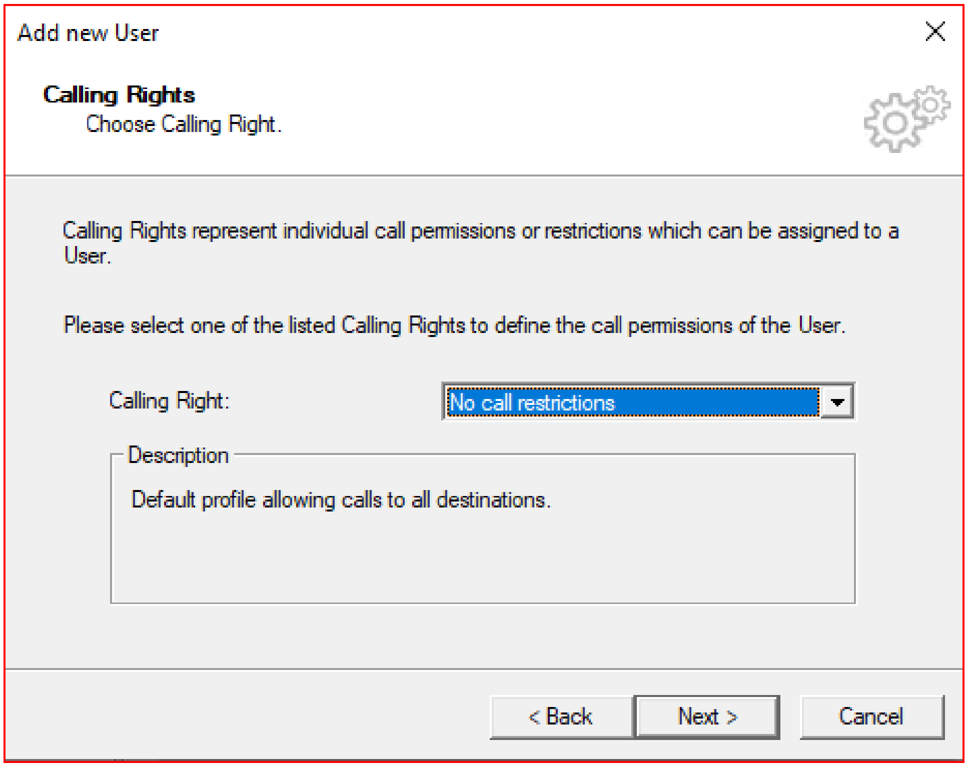
For using SwyxPhone Lxxx a PIN is required. Click on 'Create PIN' for assigning a new, unique PIN to the User.

Please inform the User about the created PIN.

You can change the PIN later on the User's 'Administration' property page.

SwyxPhone Lxxx PIN: 222222

10. Select a Calling Right and then click Next.



Add new User

Calling Rights

Choose Calling Right.

Calling Rights represent individual call permissions or restrictions which can be assigned to a User.

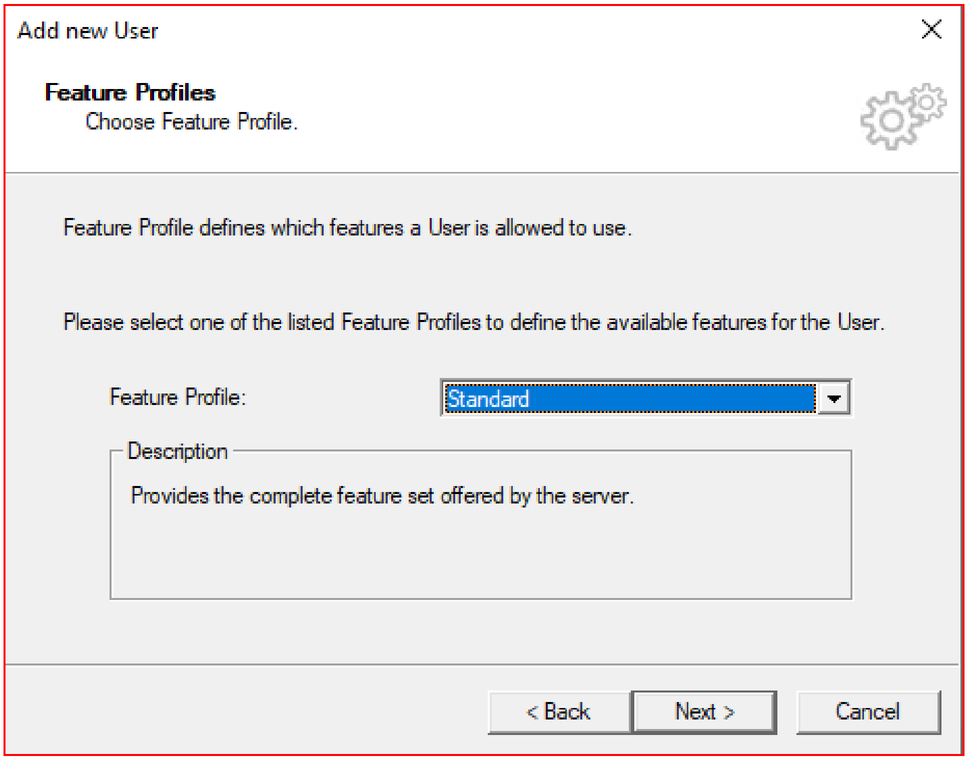
Please select one of the listed Calling Rights to define the call permissions of the User.

Calling Right: No call restrictions

Description

Default profile allowing calls to all destinations.

11. Select a Feature profile and click Next.



Add new User

Feature Profiles

Choose Feature Profile.

Feature Profile defines which features a User is allowed to use.

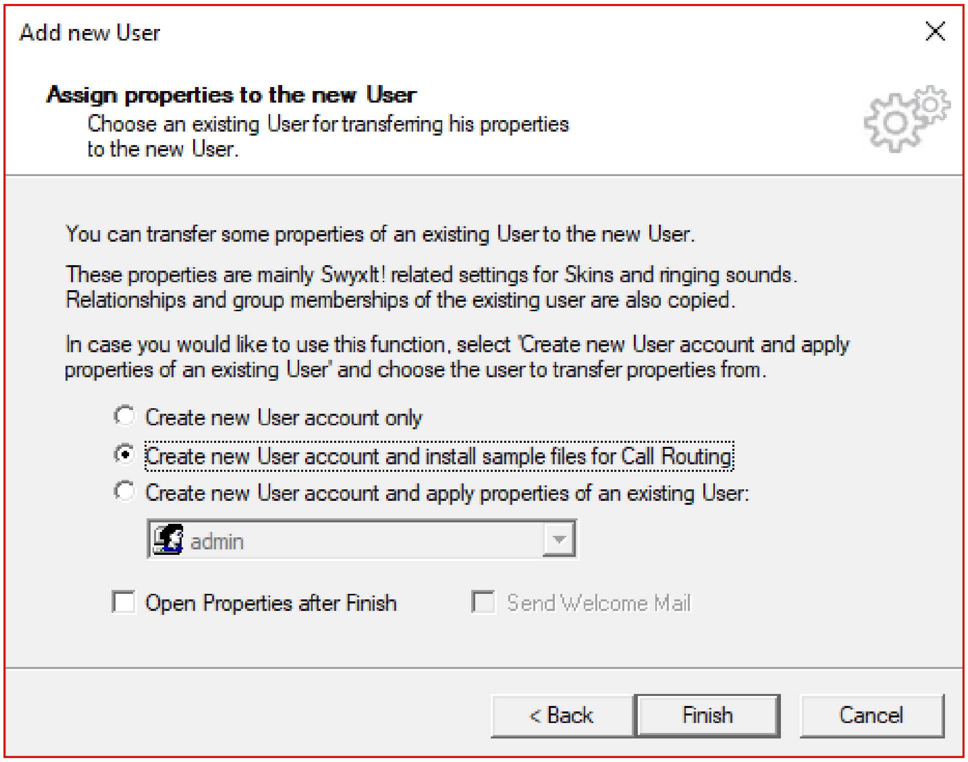
Please select one of the listed Feature Profiles to define the available features for the User.

Feature Profile: Standard

Description

Provides the complete feature set offered by the server.

12. Assign properties to the new user and click Finish.



Add new User

Assign properties to the new User

Choose an existing User for transfering his properties to the new User.

You can transfer some properties of an existing User to the new User.

These properties are mainly SwyxIt! related settings for Skins and ringing sounds. Relationships and group memberships of the existing user are also copied.

In case you would like to use this function, select 'Create new User account and apply properties of an existing User' and choose the user to transfer properties from.

○ Create new User account only

● Create new User account and install sample files for Call Routing

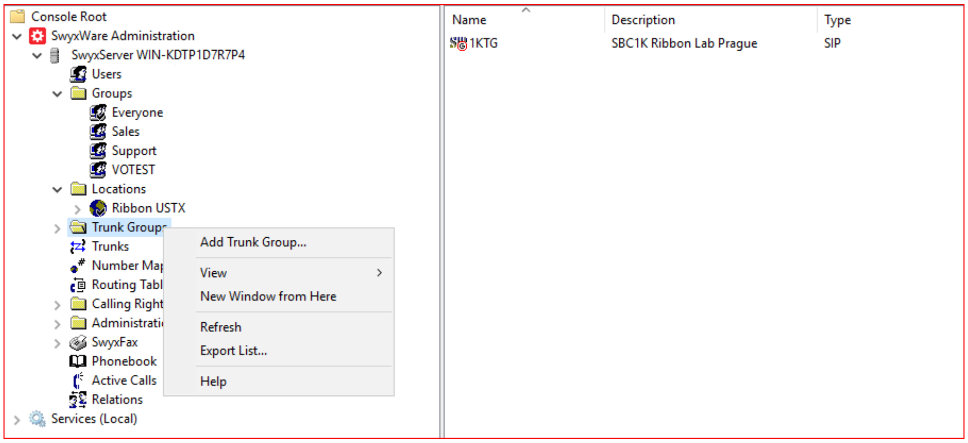
○ Create new User account and apply properties of an existing User:

admin

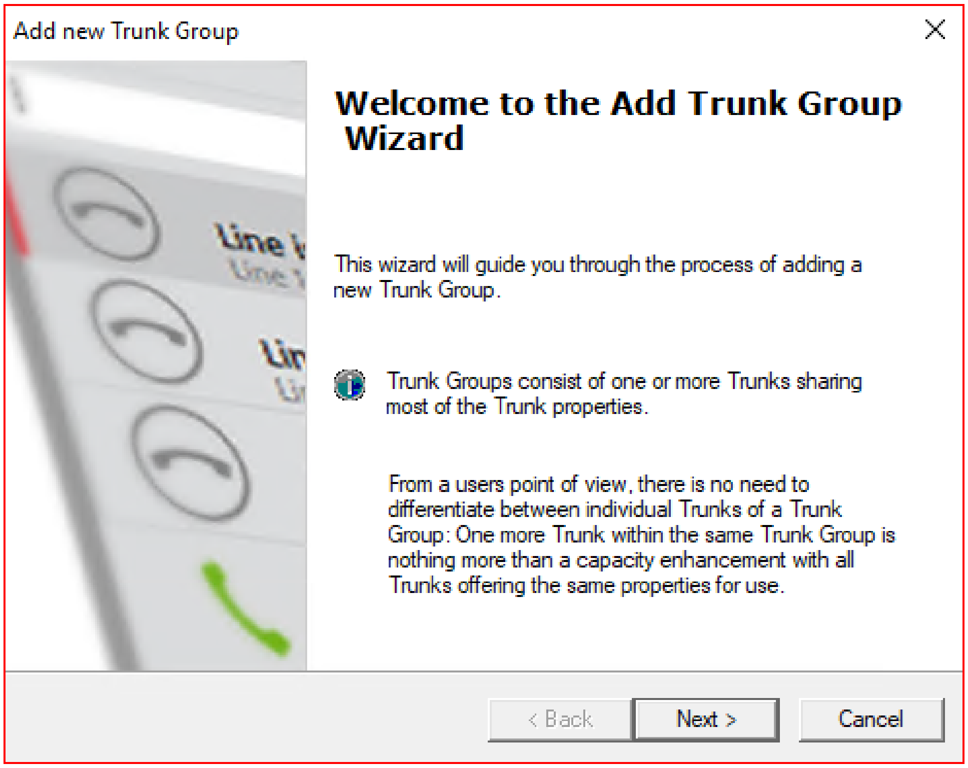
□ Open Properties after Finish   □ Send Welcome Mail

Configuring a SIP Trunk

1. Right Click on Trunk Group and select Add Trunk Group The Add Trunk Group Wizard pops up then click Next.



Name	Description	Type
1KTG	SBC1K Ribbon Lab Prague	SIP


Add new Trunk Group

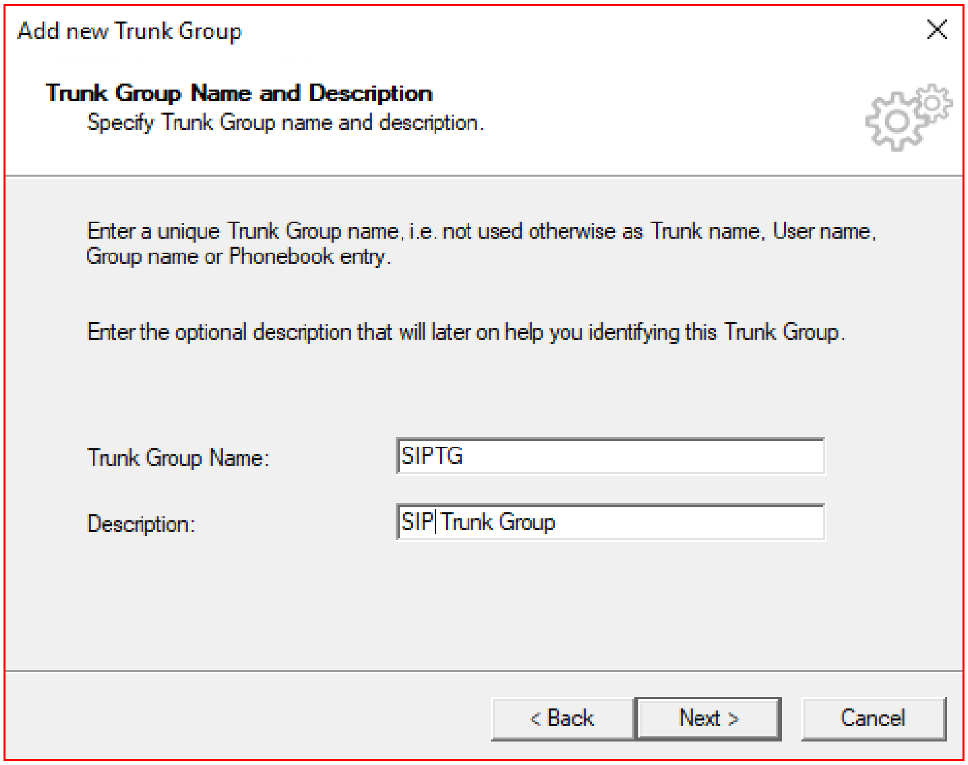
Welcome to the Add Trunk Group Wizard

This wizard will guide you through the process of adding a new Trunk Group.

Trunk Groups consist of one or more Trunks sharing most of the Trunk properties.

From a users point of view, there is no need to differentiate between individual Trunks of a Trunk Group: One more Trunk within the same Trunk Group is nothing more than a capacity enhancement with all Trunks offering the same properties for use.

2. Add the Trunk Group Name and Description and click Next.



Add new Trunk Group

Trunk Group Name and Description

Specify Trunk Group name and description.

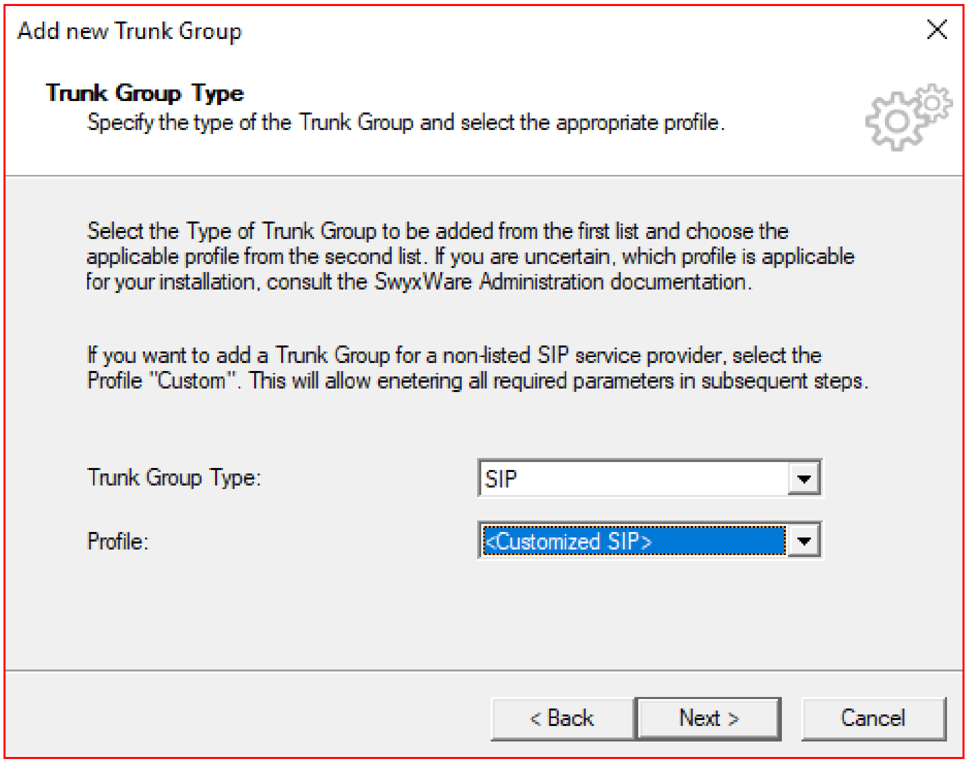
Enter a unique Trunk Group name, i.e. not used otherwise as Trunk name, User name, Group name or Phonebook entry.

Enter the optional description that will later on help you identifying this Trunk Group.

Trunk Group Name: SIPTG

Description: SIP Trunk Group

3. Select the Trunk Group Type and click Next.



Add new Trunk Group

Trunk Group Type

Specify the type of the Trunk Group and select the appropriate profile.

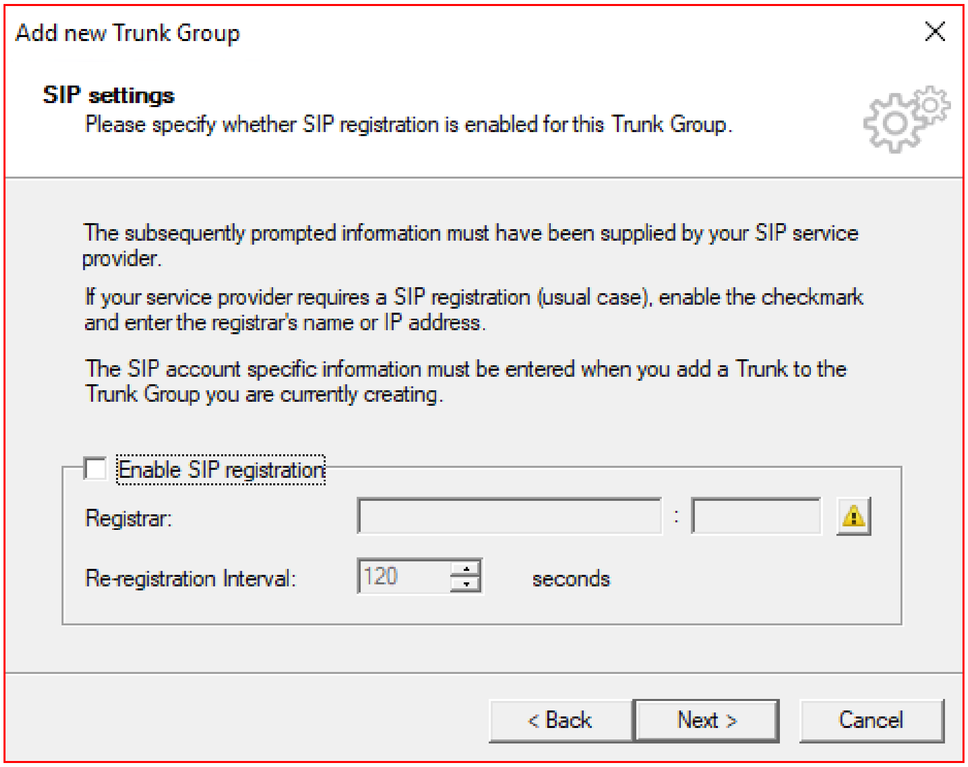
Select the Type of Trunk Group to be added from the first list and choose the applicable profile from the second list. If you are uncertain, which profile is applicable for your installation, consult the SwyxWare Administration documentation.

If you want to add a Trunk Group for a non-listed SIP service provider, select the Profile "Custom". This will allow enetering all required parameters in subsequent steps.

Trunk Group Type: SIP

Profile: <Customized SIP>

4. Add SIP settings and click Next.



Add new Trunk Group

SIP settings

Please specify whether SIP registration is enabled for this Trunk Group.

The subsequently prompted information must have been supplied by your SIP service provider.

If your service provider requires a SIP registration (usual case), enable the checkmark and enter the registrar's name or IP address.

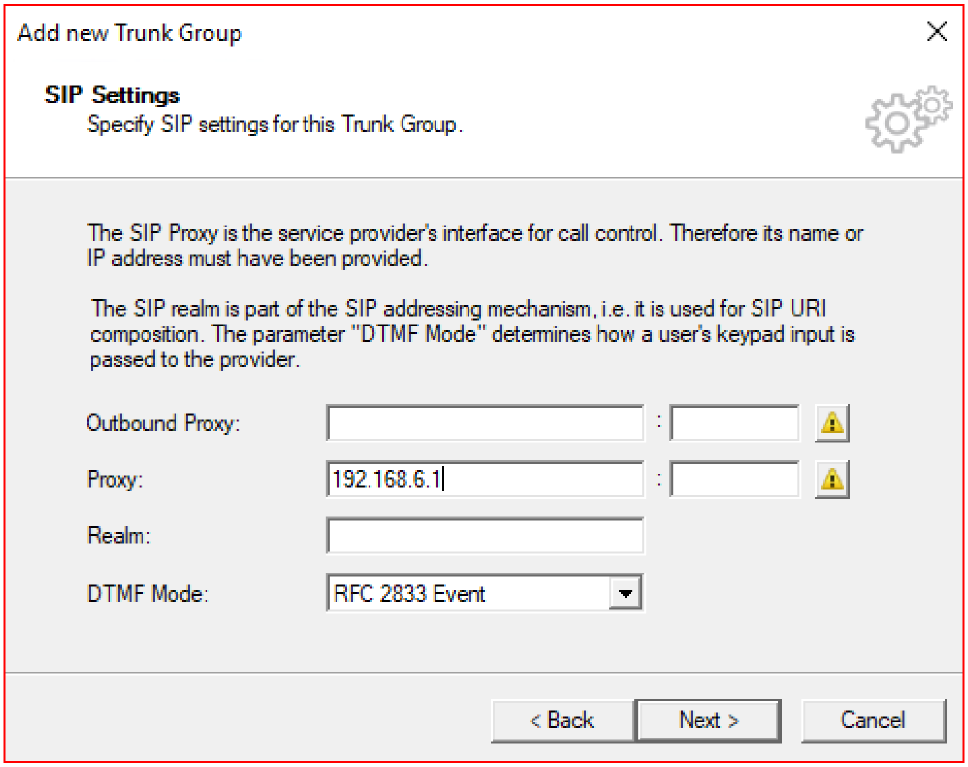
The SIP account specific information must be entered when you add a Trunk to the Trunk Group you are currently creating.

☐ Enable SIP registration

Registrar: :

Re-registration Interval: 120 seconds

5. Add the SIP Proxy and leave SIP port blank (auto-resolves) and click Next.



Add new Trunk Group

SIP Settings

Specify SIP settings for this Trunk Group.

The SIP Proxy is the service provider's interface for call control. Therefore its name or IP address must have been provided.

The SIP realm is part of the SIP addressing mechanism, i.e. it is used for SIP URI composition. The parameter "DTMF Mode" determines how a user's keypad input is passed to the provider.

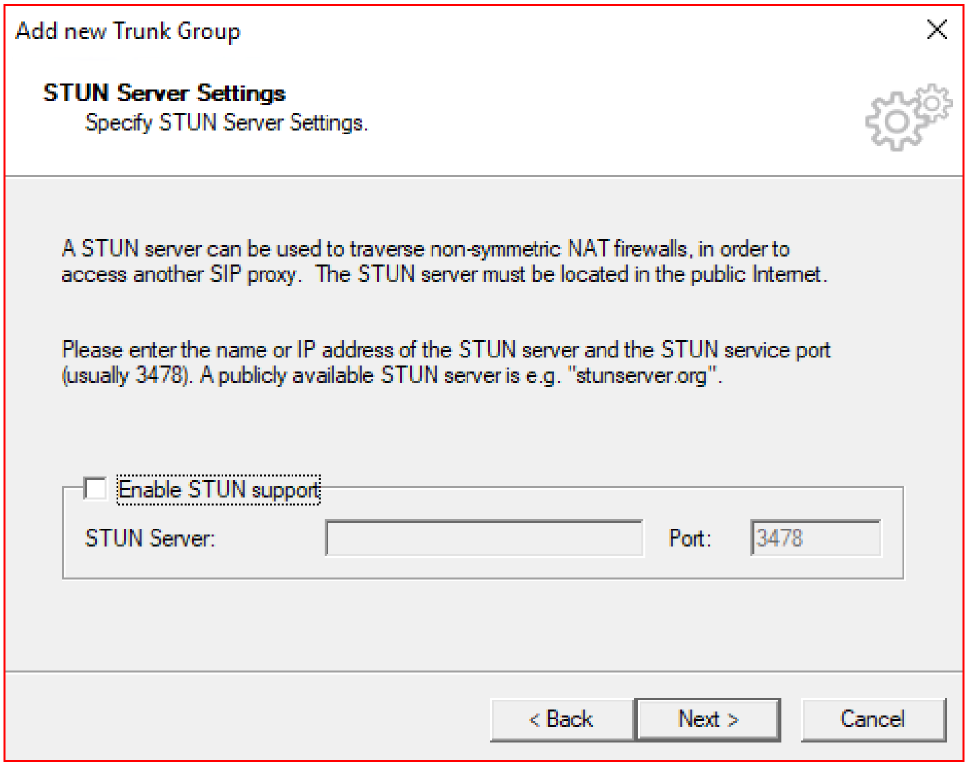
Outbound Proxy: :

Proxy: 192.168.6.1 :

Realm:

DTMF Mode: RFC 2833 Event

6. If a STUN server is supported, click the check box and add the IP for the STUN server and click Next.



Add new Trunk Group

STUN Server Settings

Specify STUN Server Settings.

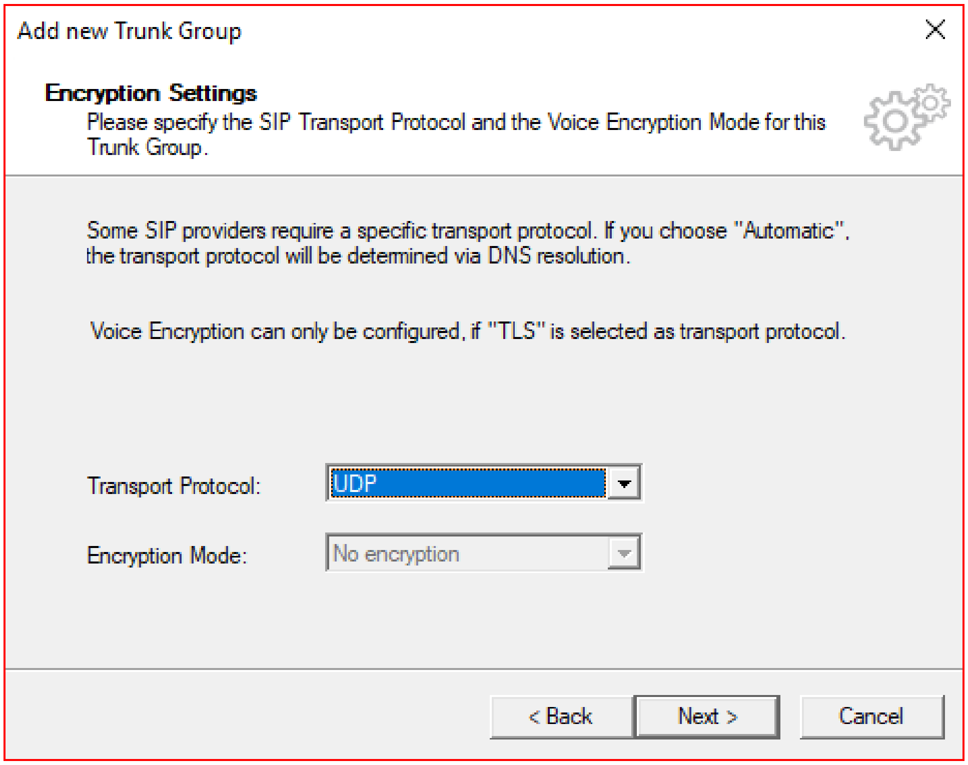
A STUN server can be used to traverse non-symmetric NAT firewalls, in order to access another SIP proxy. The STUN server must be located in the public Internet.

Please enter the name or IP address of the STUN server and the STUN service port (usually 3478). A publicly available STUN server is e.g. "stunserver.org".

☐ Enable STUN support

STUN Server: Port: 3478

7. Select Transport Protocol and click Next.



Add new Trunk Group

Encryption Settings

Please specify the SIP Transport Protocol and the Voice Encryption Mode for this Trunk Group.

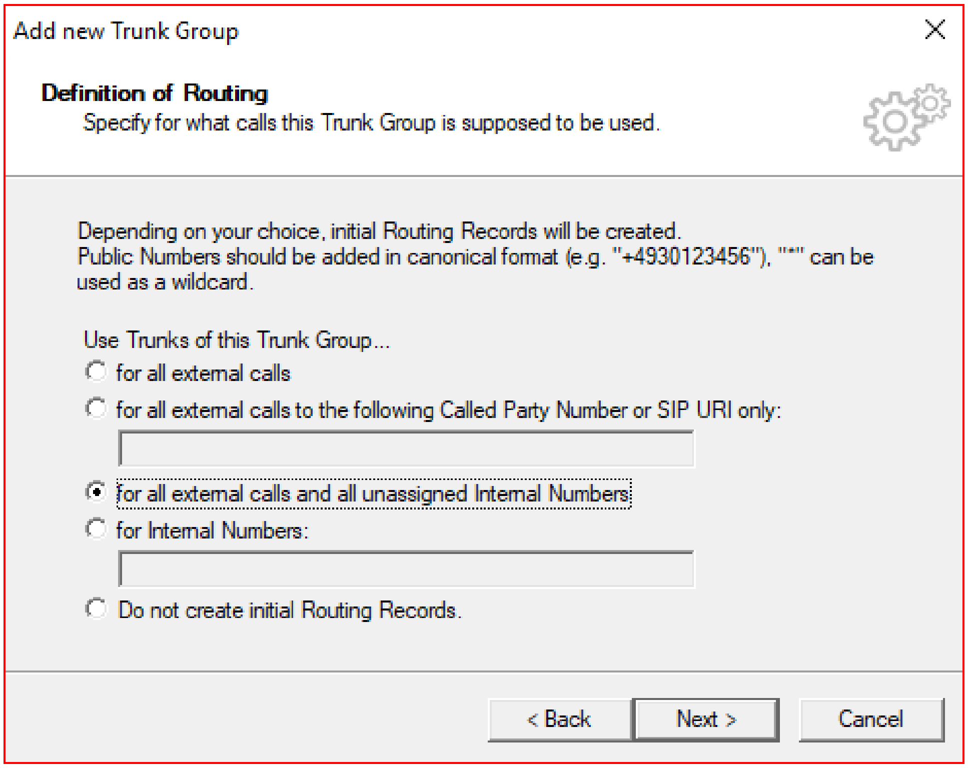
Some SIP providers require a specific transport protocol. If you choose "Automatic", the transport protocol will be determined via DNS resolution.

Voice Encryption can only be configured, if "TLS" is selected as transport protocol.

Transport Protocol: UDP

Encryption Mode: No encryption

8. Select the Routing Definition and click Next.



Add new Trunk Group

Definition of Routing

Specify for what calls this Trunk Group is supposed to be used.

Depending on your choice, initial Routing Records will be created.
Public Numbers should be added in canonical format (e.g. "+4930123456"), "*" can be used as a wildcard.

Use Trunks of this Trunk Group...

○ for all external calls

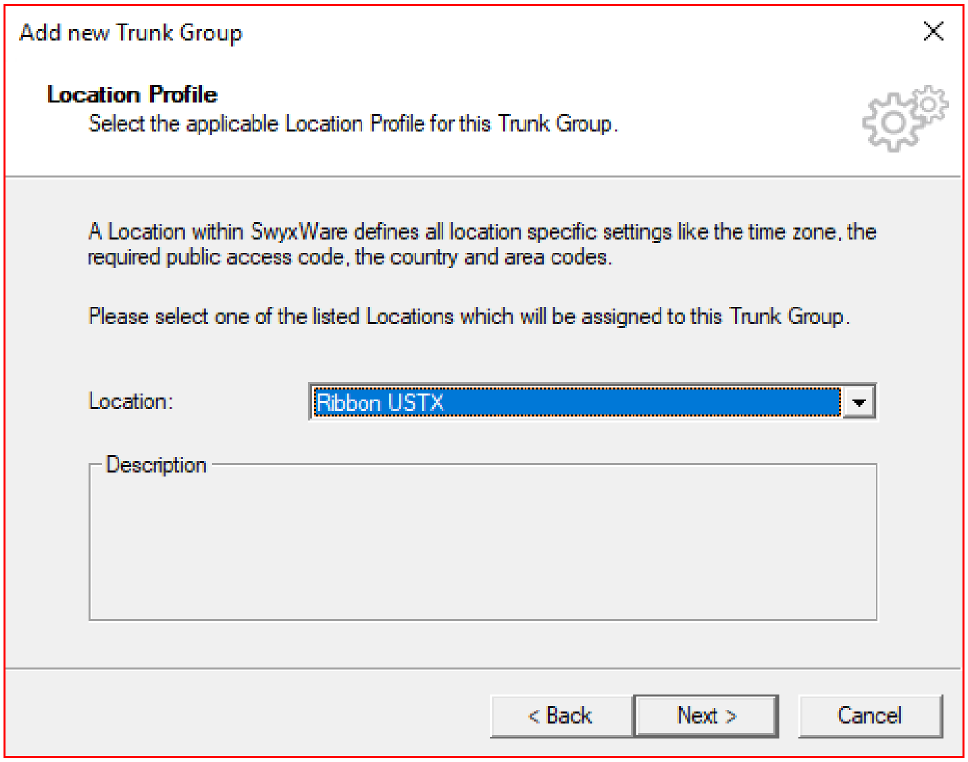
○ for all external calls to the following Called Party Number or SIP URI only:

● for all external calls and all unassigned Internal Numbers

○ for Internal Numbers:

○ Do not create initial Routing Records.

9. Select Location Profile and click Next.



Add new Trunk Group

Location Profile

Select the applicable Location Profile for this Trunk Group.

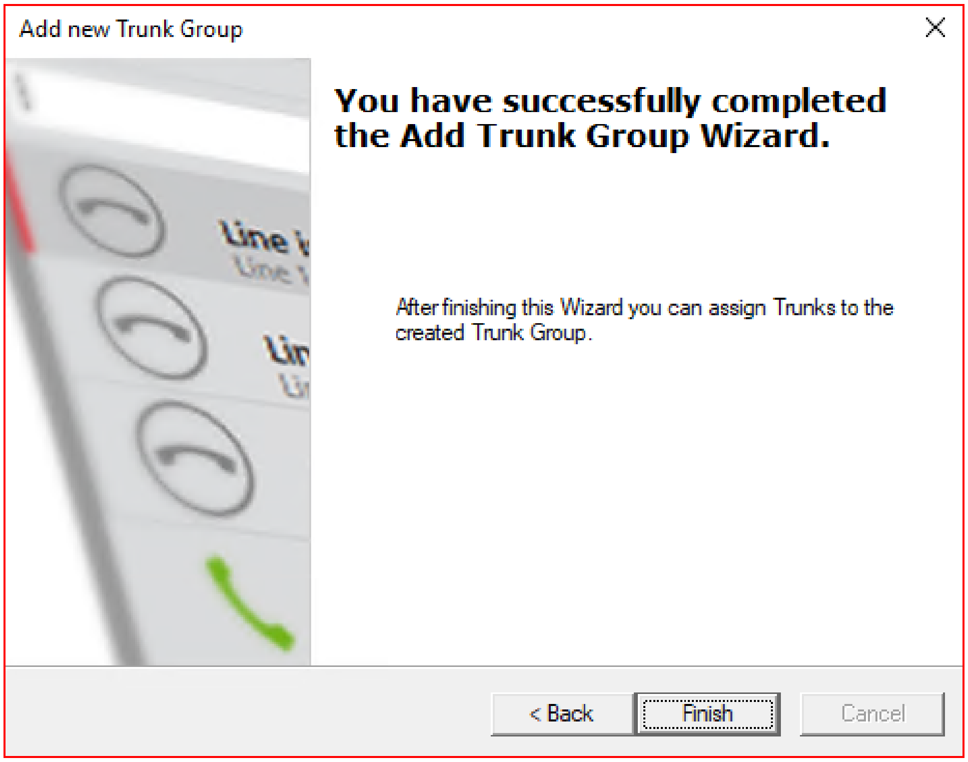
A Location within SwyxWare defines all location specific settings like the time zone, the required public access code, the country and area codes.

Please select one of the listed Locations which will be assigned to this Trunk Group.

Location: Ribbon USTX

Description

10. Click Finish.

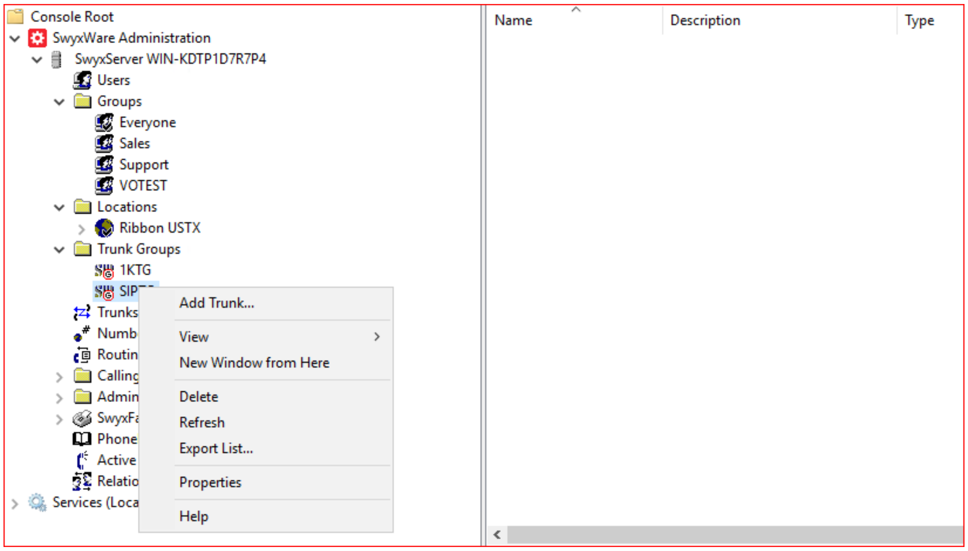


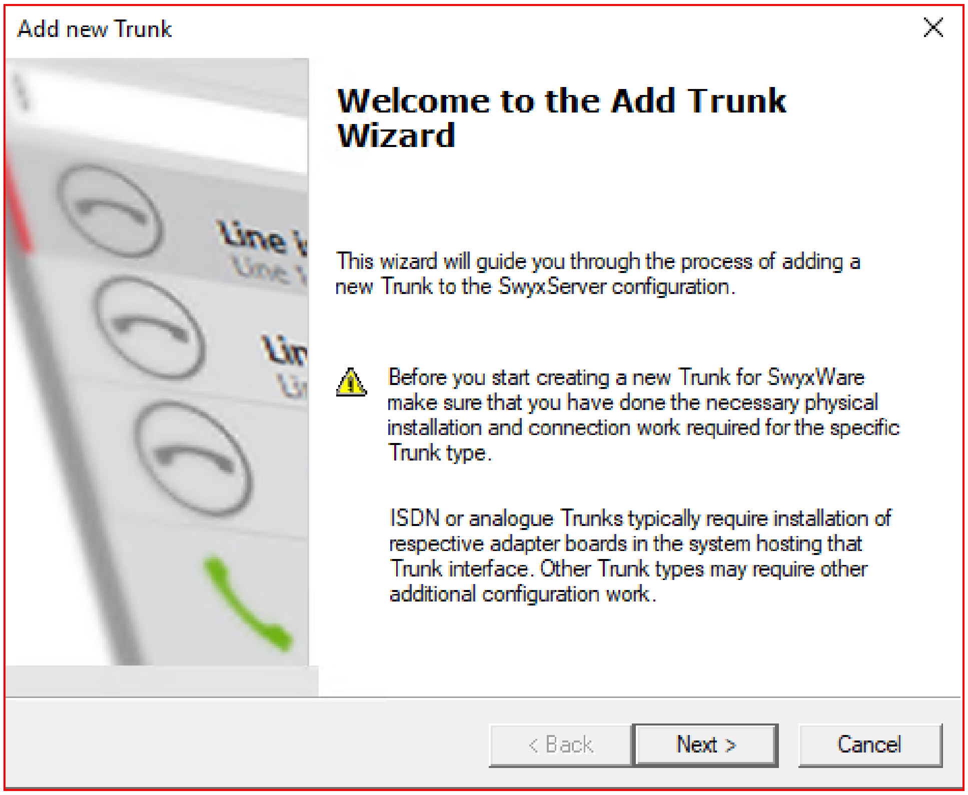
Add new Trunk Group

You have successfully completed the Add Trunk Group Wizard.

After finishing this Wizard you can assign Trunks to the created Trunk Group.

11. Right Click on the Newly created Trunk Group and click Add Trunk, The Add Trunk Wizard pops up, then click Next.





Add new Trunk

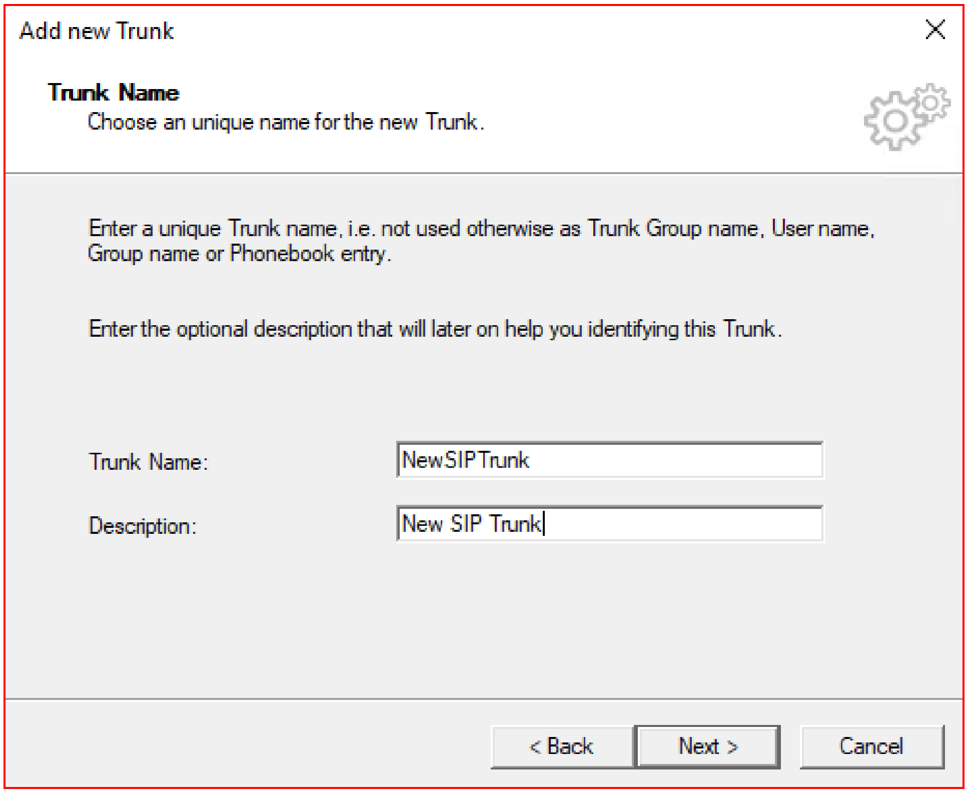
Welcome to the Add Trunk Wizard

This wizard will guide you through the process of adding a new Trunk to the SwyxServer configuration.

⚠ Before you start creating a new Trunk for SwyxWare make sure that you have done the necessary physical installation and connection work required for the specific Trunk type.

ISDN or analogue Trunks typically require installation of respective adapter boards in the system hosting that Trunk interface. Other Trunk types may require other additional configuration work.

12. Add a Trunk Name and Description.



Add new Trunk

Trunk Name

Choose an unique name for the new Trunk.

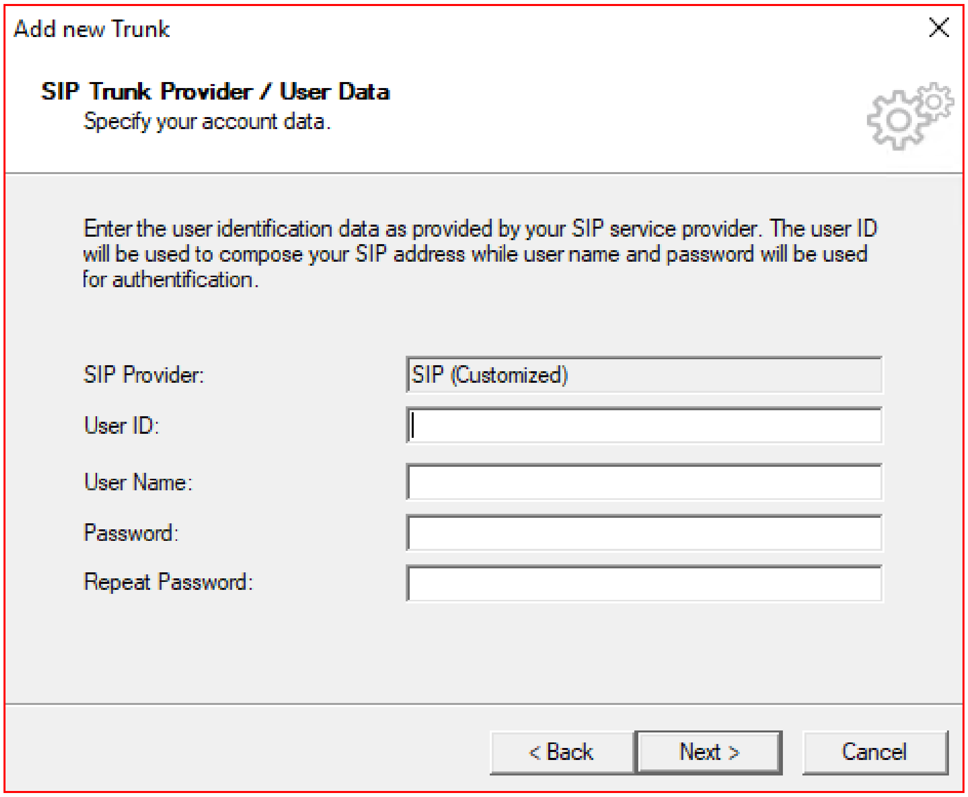
Enter a unique Trunk name, i.e. not used otherwise as Trunk Group name, User name, Group name or Phonebook entry.

Enter the optional description that will later on help you identifying this Trunk.

Trunk Name: NewSIPTrunk

Description: New SIP Trunk

13. Add the SIP trunk Provider and User Data then click Next.



Add new Trunk

SIP Trunk Provider / User Data

Specify your account data.

Enter the user identification data as provided by your SIP service provider. The user ID will be used to compose your SIP address while user name and password will be used for authentification.

SIP Provider: SIP (Customized)

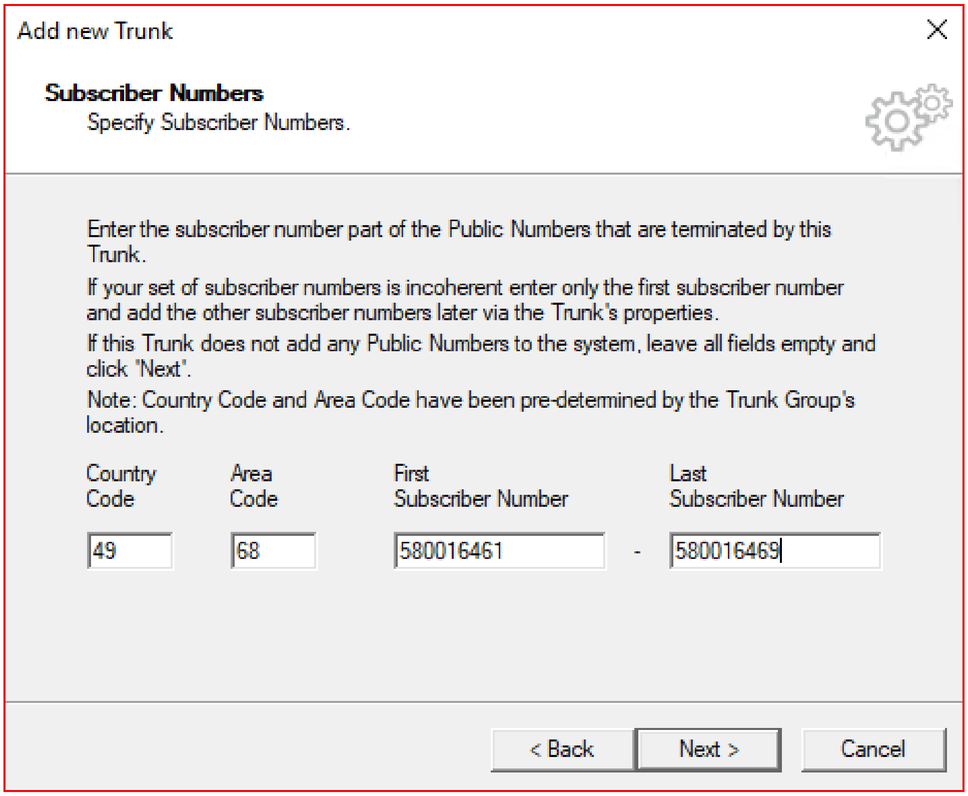
User ID:

User Name:

Password:

Repeat Password:

14. Select the Subscriber Number using the SIP Trunk and click Next.



Add new Trunk

Subscriber Numbers

Specify Subscriber Numbers.

Enter the subscriber number part of the Public Numbers that are terminated by this Trunk.

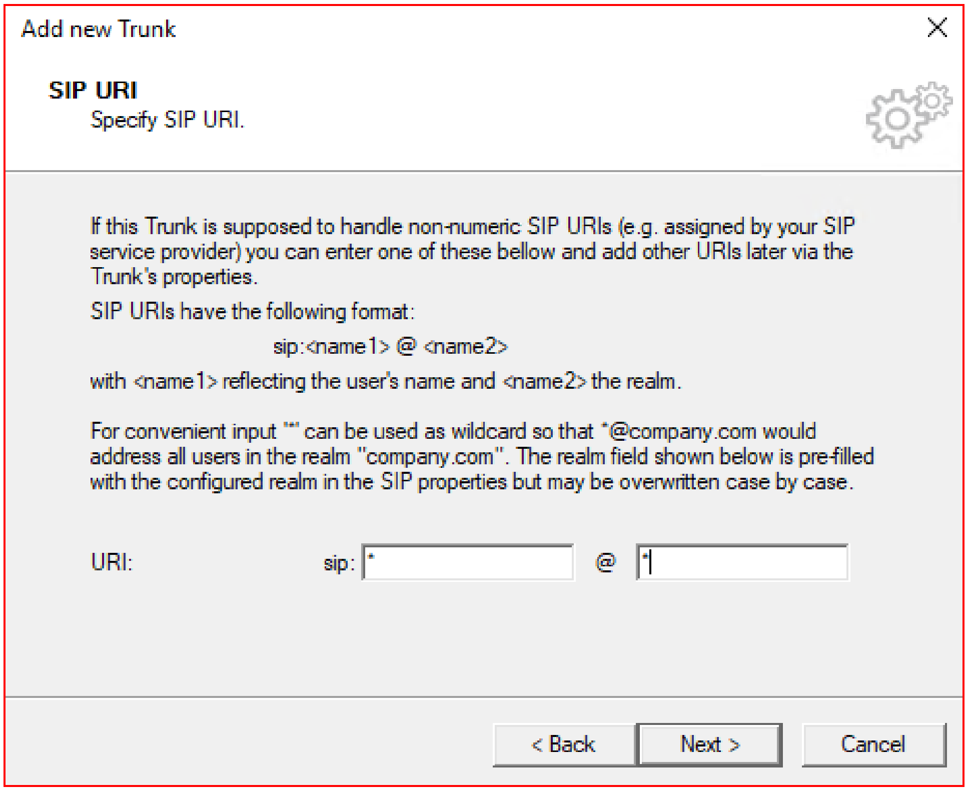
If your set of subscriber numbers is incoherent enter only the first subscriber number and add the other subscriber numbers later via the Trunk's properties.

If this Trunk does not add any Public Numbers to the system, leave all fields empty and click "Next".

Note: Country Code and Area Code have been pre-determined by the Trunk Group's location.

Country Code	Area Code	First Subscriber Number		Last Subscriber Number
49	68	580016461	-	580016469

15. Add a SIP URI (wild card "*" for any) and then click Next.



Add new Trunk

SIP URI

Specify SIP URI.

If this Trunk is supposed to handle non-numeric SIP URIs (e.g. assigned by your SIP service provider) you can enter one of these bellow and add other URIs later via the Trunk's properties.

SIP URIs have the following format:

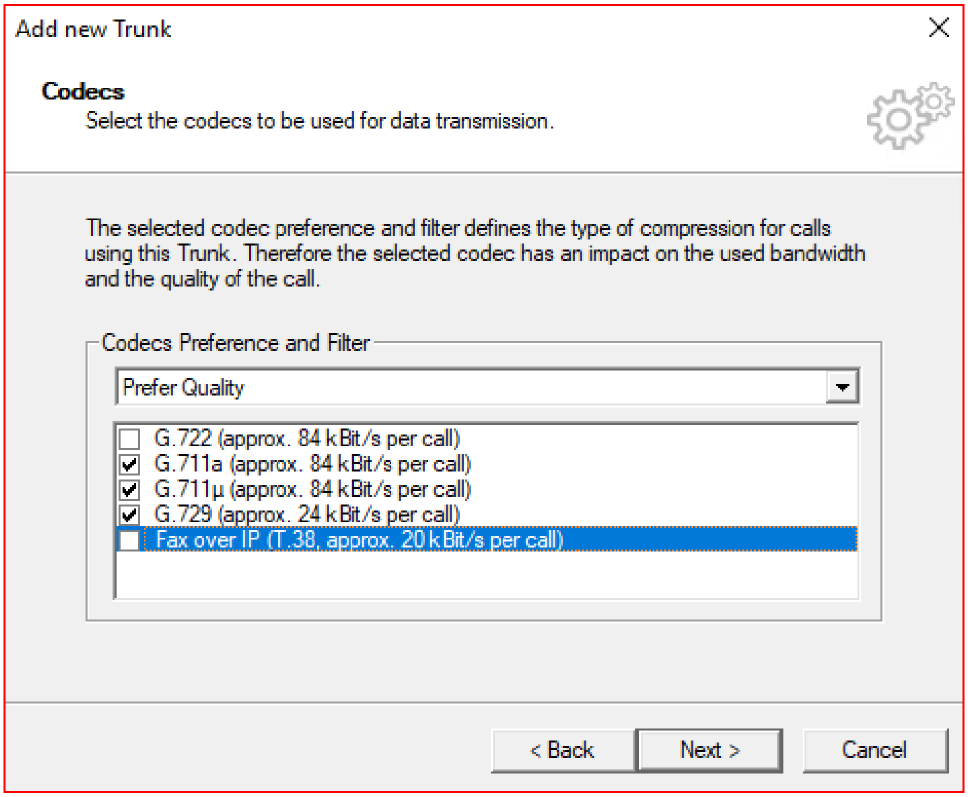
sip:<name1> @ <name2>

with <name1> reflecting the user's name and <name2> the realm.

For convenient input "*" can be used as wildcard so that *@company.com would address all users in the realm "company.com". The realm field shown below is pre-filled with the configured realm in the SIP properties but may be overwritten case by case.

URI: sip: * @ *

16. Select the Codecs supported by the SIP trunk and click Next.



Add new Trunk

Codecs

Select the codecs to be used for data transmission.

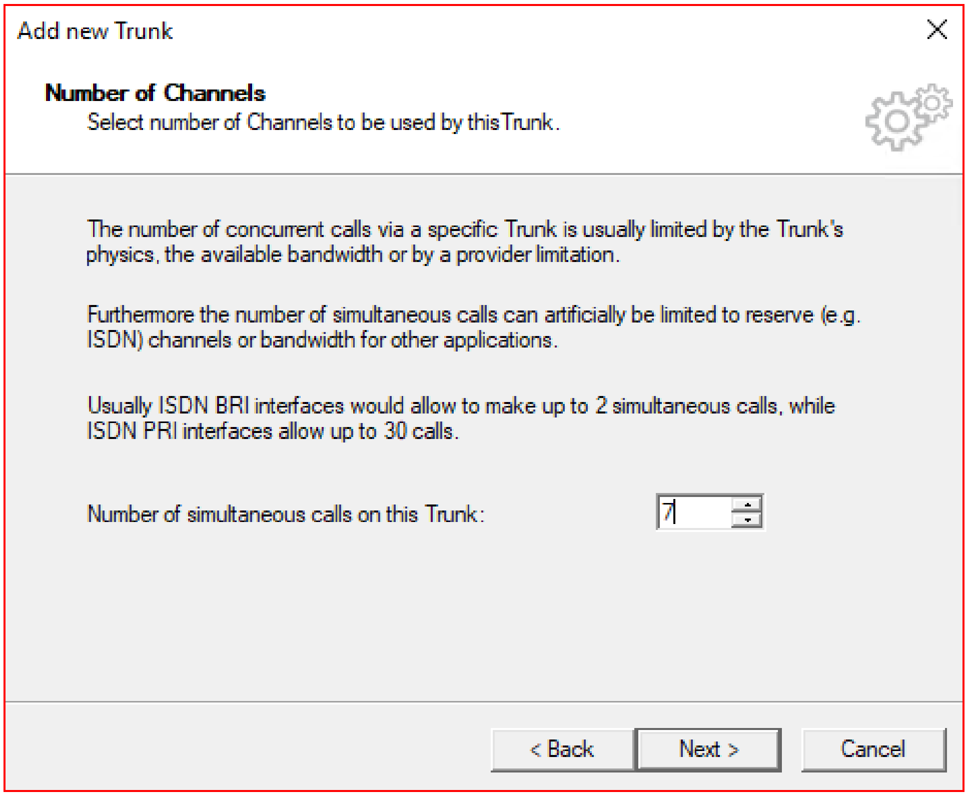
The selected codec preference and filter defines the type of compression for calls using this Trunk. Therefore the selected codec has an impact on the used bandwidth and the quality of the call.

Codecs Preference and Filter

Prefer Quality

- □ G.722 (approx. 84 kBit/s per call)
- ☑ G.711a (approx. 84 kBit/s per call)
- ☑ G.711µ (approx. 84 kBit/s per call)
- ☑ G.729 (approx. 24 kBit/s per call)
- □ Fax over IP (T.38, approx. 20 kBit/s per call)

17. Select the Number of Simultaneous calls possible in the SIP Trunk and click Next.



Add new Trunk

Number of Channels

Select number of Channels to be used by this Trunk.

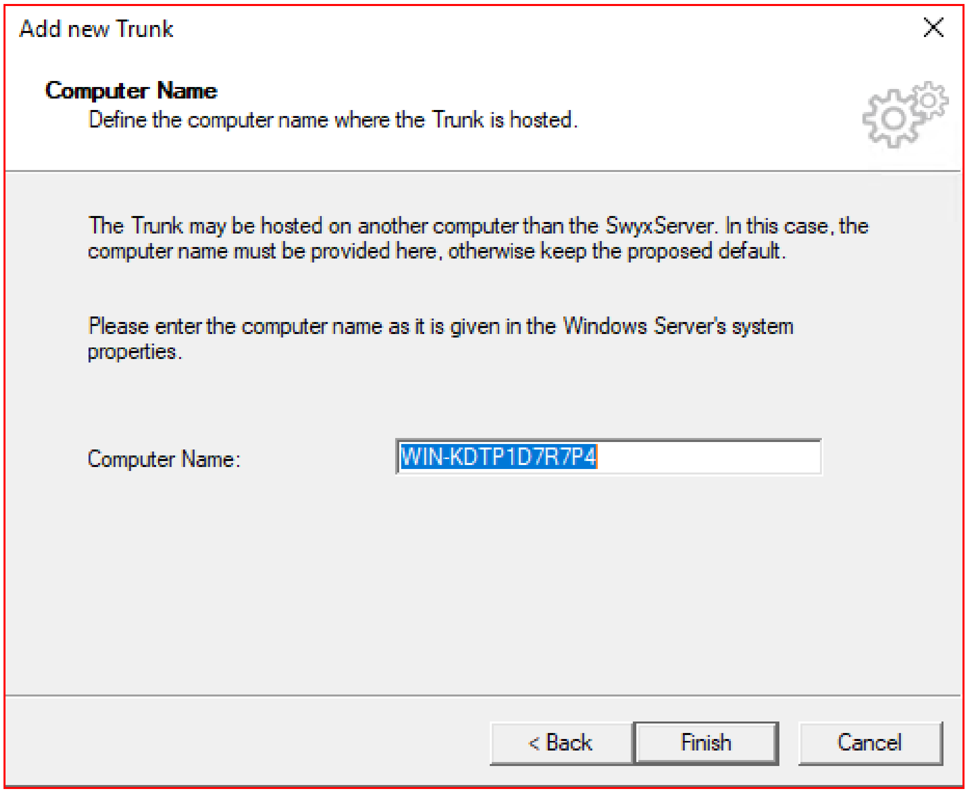
The number of concurrent calls via a specific Trunk is usually limited by the Trunk's physics, the available bandwidth or by a provider limitation.

Furthermore the number of simultaneous calls can artificially be limited to reserve (e.g. ISDN) channels or bandwidth for other applications.

Usually ISDN BRI interfaces would allow to make up to 2 simultaneous calls, while ISDN PRI interfaces allow up to 30 calls.

Number of simultaneous calls on this Trunk: 7

18. Choose PSX server or Computer Name and click Finish.



Add new Trunk

Computer Name

Define the computer name where the Trunk is hosted.

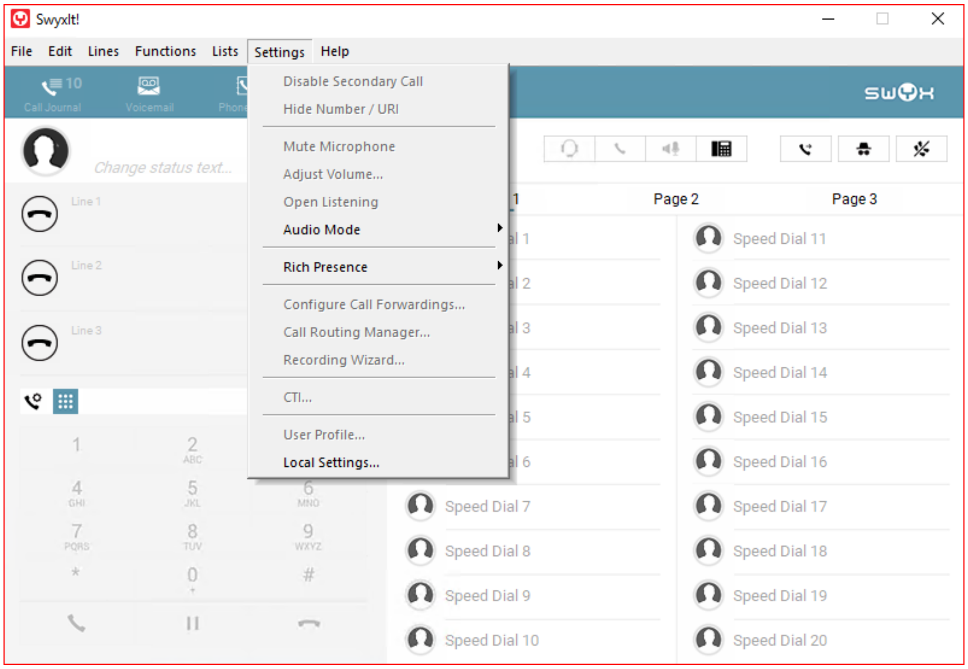
The Trunk may be hosted on another computer than the SwyxServer. In this case, the computer name must be provided here, otherwise keep the proposed default.

Please enter the computer name as it is given in the Windows Server's system properties.

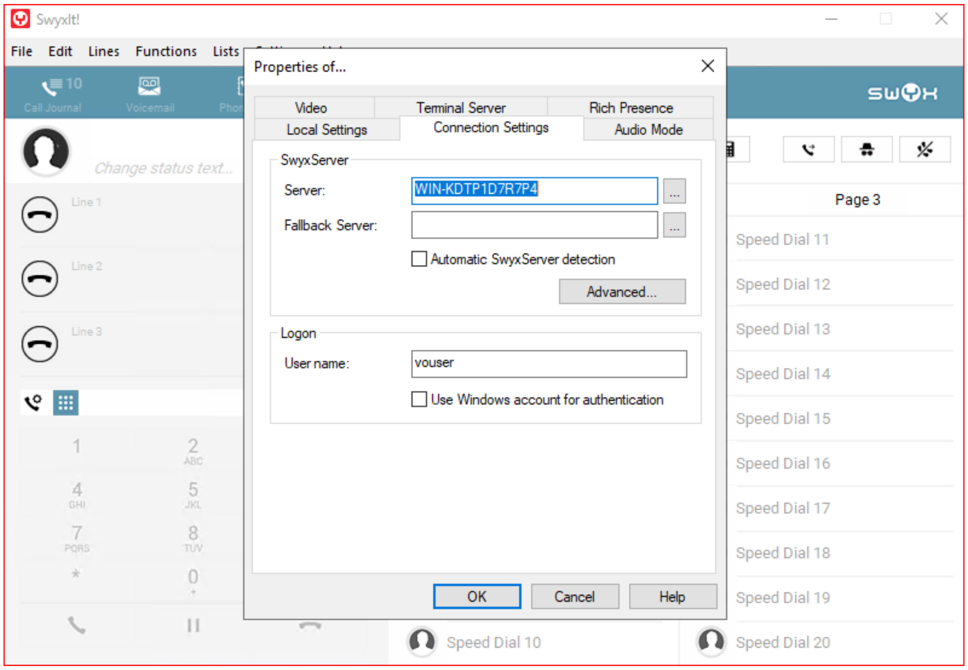
Computer Name: WIN-KDTP1D7R7P4

Configuring the SwyxIt Client

1. Click On Settings and select Local Settings



2. Click the Connections Settings Tab and add the Server name or IP address and the User Name, then click OK.



Properties of...

SwyxServer

Server: WIN-KDTP1D7R7P4

Fallback Server:

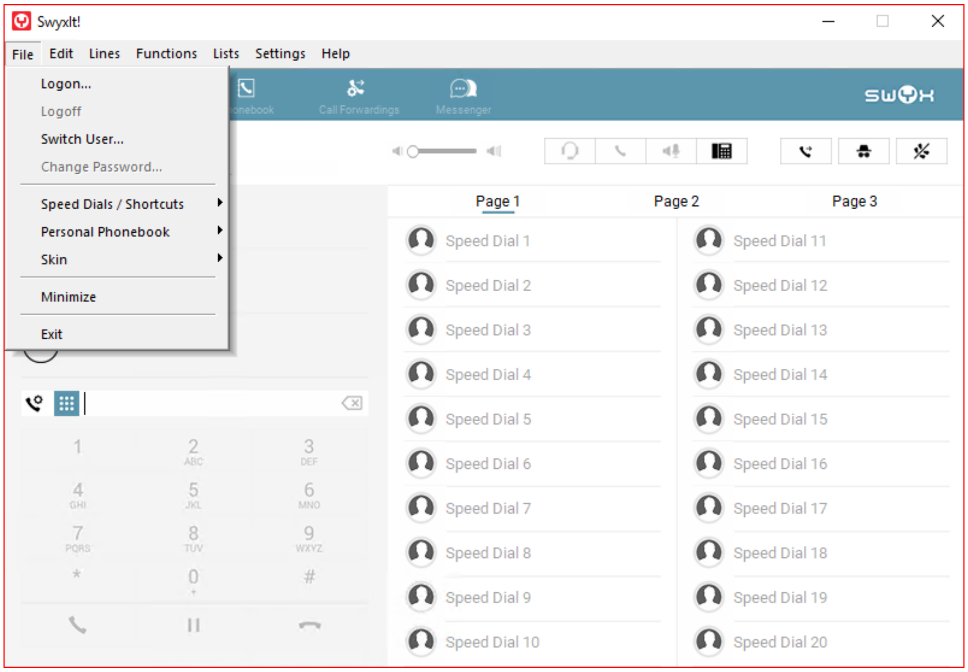
□ Automatic SwyxServer detection

Logon

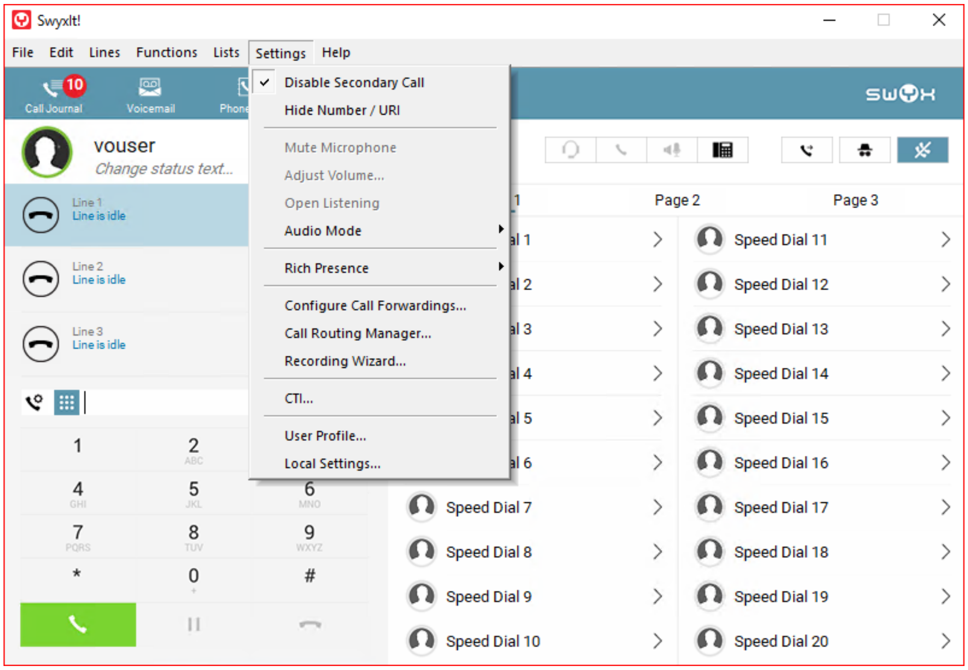
User name: vouser

□ Use Windows account for authentication

3. From the File menu select Logon.



4. Once Logged in, More Choices are available in the Settings Menu. Some configuration capability is also available from the User Properties on the PBX server.

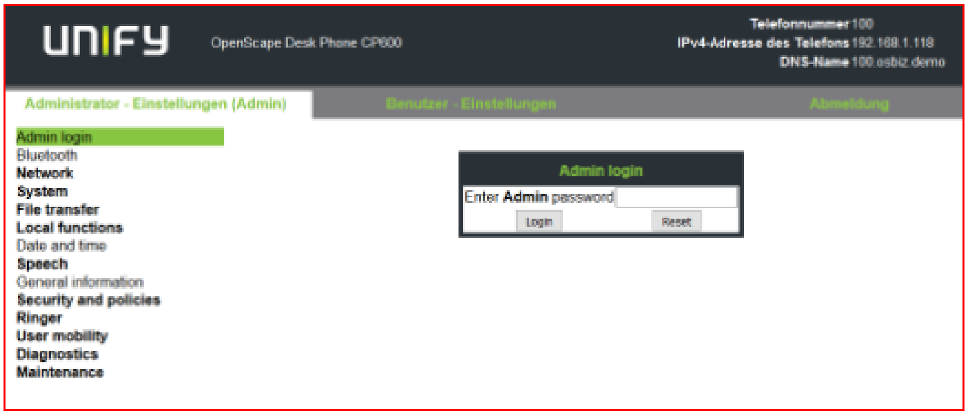


Configuring the HFA Phone

The Unify CP 205 comes out of the box as a SIP Phone. You must upgrade the firmware before configuring HFA Phone. Upgrade the firmware using FTP/HTTPS Access Data

By default, the phone has DHCP enabled. Look on the EdgeMarc 2900 for the IP leased to the CP205 unit.

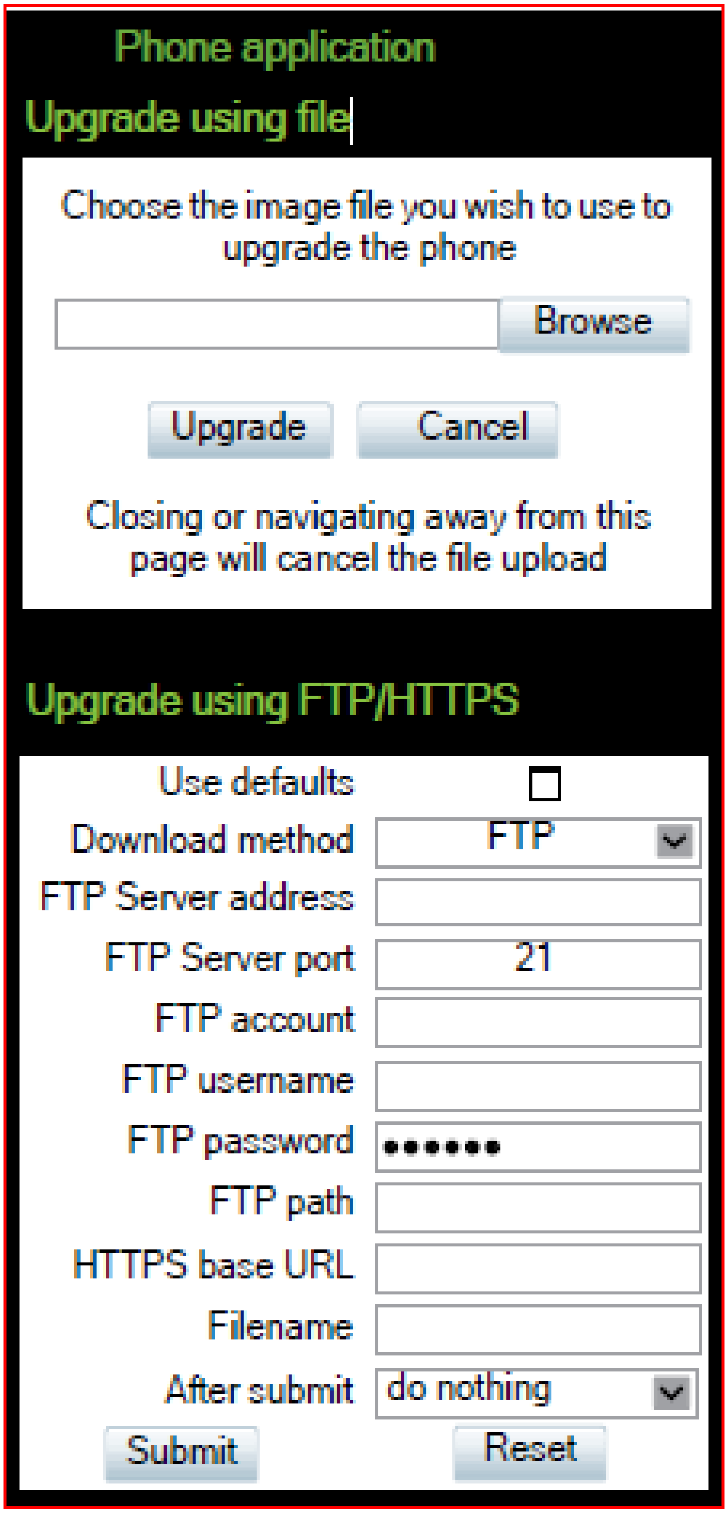
1. Open your web browser and enter the appropriate URL. Example: https://192.168.1.15.


Telefonnummer 100
IPv4-Adresse des Telefons 192.168.1.118
DNS-Name 100.osbiz.demo

Note: the default password is 123456.

2. From the Administration via Web-Based Management (WBM), select File transfer > Phone application.

   Click Browse to find the corresponding L62.iso file to upload to the Unify CP205 Phones. Then, click Submit.



Phone application

Upgrade using file

Choose the image file you wish to use to upgrade the phone

[ ] Browse

Upgrade Cancel

Closing or navigating away from this page will cancel the file upload

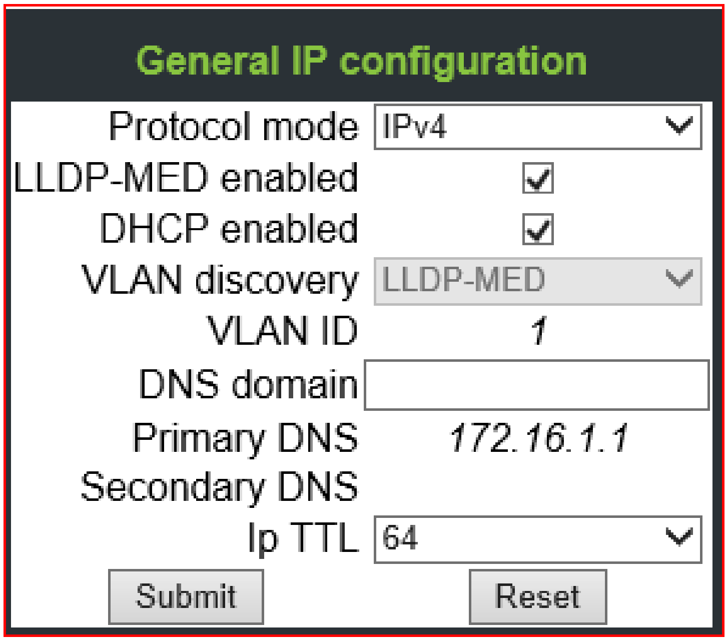
Upgrade using FTP/HTTPS

Use defaults	□
Download method	FTP
FTP Server address	
FTP Server port	21
FTP account	
FTP username	
FTP password	••••••
FTP path	
HTTPS base URL	
Filename	
After submit	do nothing

Submit Reset

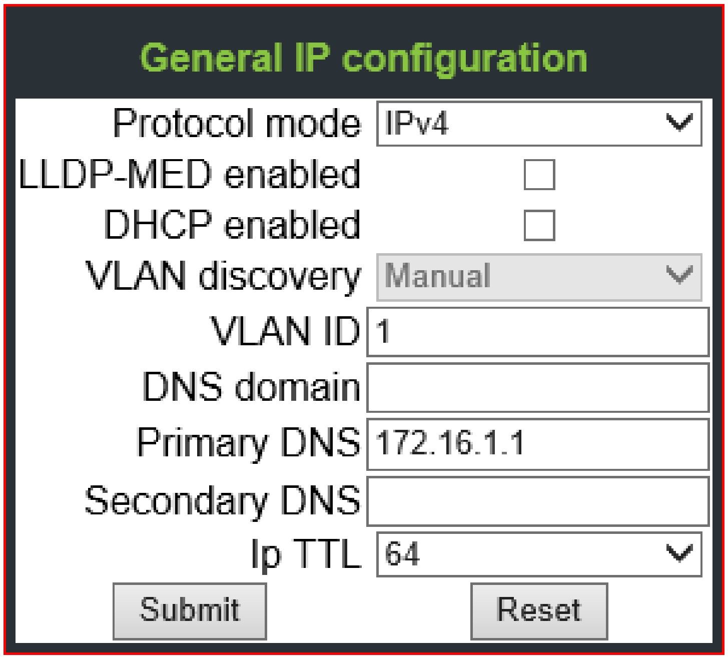
3. Once the firmware has been upgraded Login to Web-Based Management to configure the network settings by selecting Network > General IP configuration, then click Submit.

   Choose between DHCP or Static IP assignment.



General IP configuration

Protocol mode	IPv4
LLDP-MED enabled	☑
DHCP enabled	☑
VLAN discovery	LLDP-MED
VLAN ID	*1*
DNS domain	
Primary DNS	*172.16.1.1*
Secondary DNS	
Ip TTL	64

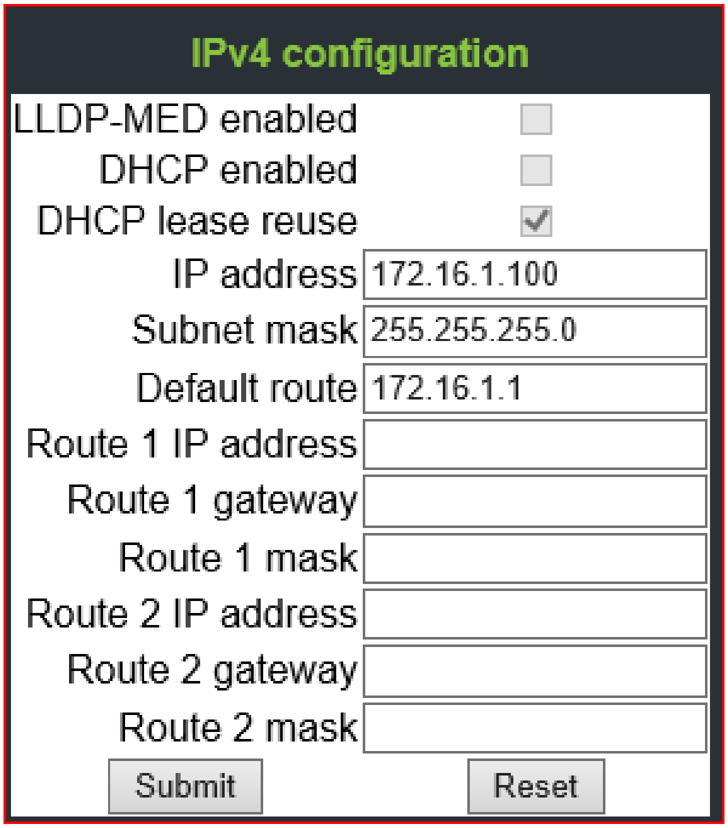


General IP configuration

Protocol mode	IPv4
LLDP-MED enabled	□
DHCP enabled	□
VLAN discovery	Manual
VLAN ID	1
DNS domain	
Primary DNS	172.16.1.1
Secondary DNS	
Ip TTL	64

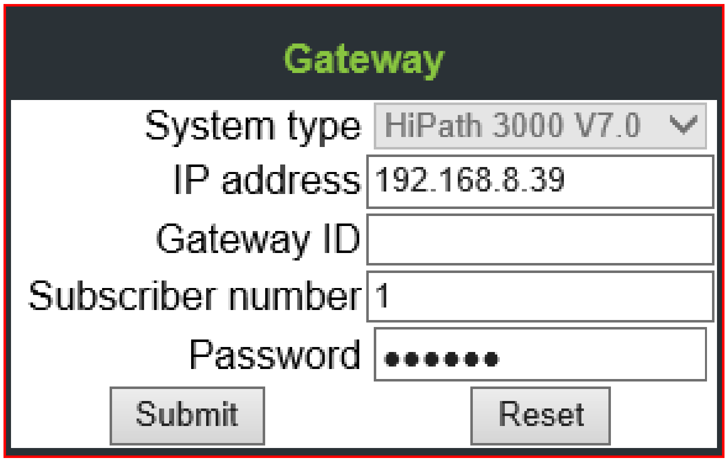
4. Select Network > IPV4 configuration, then click Submit. Add IP address, Subnet Mask, and Default Route.

   Note: If DHCP is enabled the values will be fetched automatically.



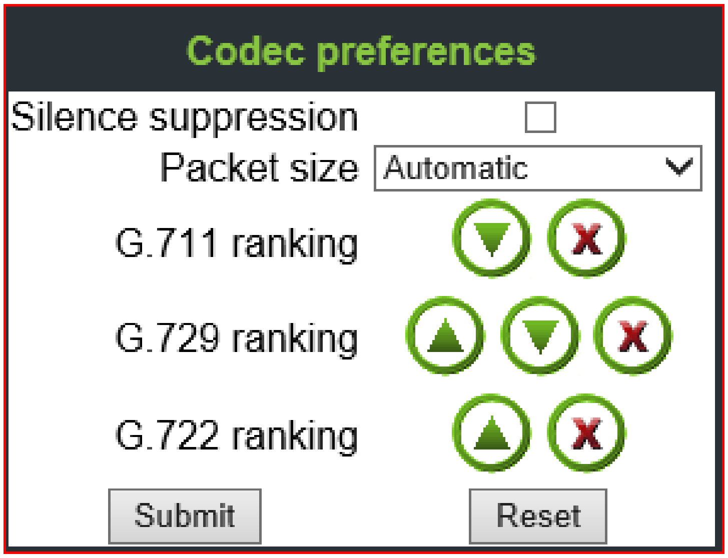
LLDP-MED enabled	☐
DHCP enabled	☐
DHCP lease reuse	☑
IP address	172.16.1.100
Subnet mask	255.255.255.0
Default route	172.16.1.1
Route 1 IP address	
Route 1 gateway	
Route 1 mask	
Route 2 IP address	
Route 2 gateway	
Route 2 mask	

5. Add Gateway information by selecting System > Gateway, then click Submit.



System type	HiPath 3000 V7.0
IP address	192.168.8.39
Gateway ID	
Subscriber number	1
Password	••••••

6. Choose Codecs by selecting Speech > Codec preferences, then click Submit.



Codec preferences

Silence suppression □

Packet size: Automatic

G.711 ranking

G.729 ranking

G.722 ranking

7. Go to Phone and login to SwyxPBX using the user created in section Adding a User.

Supplementary Services and Features Coverage

The following checklist identifies the set of supplementary services/features covered through the configuration defined in this Interop document.

Sr. No.	Supplementary Services/Features	Coverage
1	Registration over UDP/TCP/TLS	✓
2	Basic Call Setup & Termination	✓
3	Ringing & Local Ringback Tone	✓
4	Remote Ringback Tone Handling	

		✓
5	Cancel Call & Call Rejection	✓
6	Call Forwarding Busy	✓
7	Call Forward No Answer	✓
8	Call Transfer (Attended)	✓
9	Call Transfer (Blind/ Unattended)	✓
10	Conference Call	✓

Legend

Supported	✓
Not Supported	✗

Caveats

The following items should be noted in relation to this Interop document. These are either limitations, untested elements, or useful information pertaining to the Interoperability.

- Fax calls and other test were not performed due to unavailability of required devices at the Ribbon Lab.

Support

For any support related queries about this guide, please contact your local Ribbon representative, or use the details below:

- Sales and Support: 1-833-742-2661
- Other Queries: 1-877-412-8867
- Website: https://ribboncommunications.com/about-us

References

For detailed information about Ribbon products & solutions, please visit:

https://ribboncommunications.com/products

Conclusion

This Interoperability document describes a successful configuration and interop involving EdgeMarc 2900 and SwyxWare PBX.

All features and capabilities tested are detailed within this document. Any limitations, notes or observations are also recorded to provide the reader with an accurate understanding of what has been covered, and what has not.

Configuration guidance is provided to enable the reader to replicate the same base setup - there maybe additional configuration changes required to suit the exact deployment environment.