EdgeMarc 2900 POE Interop with Vodafone SIP Trunk

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Interoperable Carrier

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Document Overview

This document outlines the configuration best practices for **Vodafone SIP Trunk** 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. "**Swyx on-Prem**" The inhouse model is based on the Swyx software being installed on a server in your company. This can be a dedicated Windows server or a virtualized server system. The administration and maintenance of the system is carried out by your IT or a certified Swyx reseller.

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 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 are 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.
- Tested with EdgeMarc VOS version 15.8.3

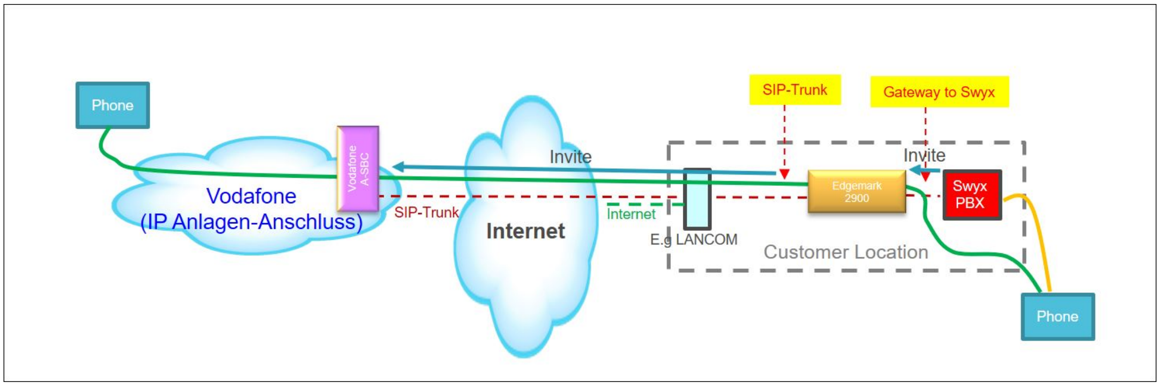
Product and Device Details

	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

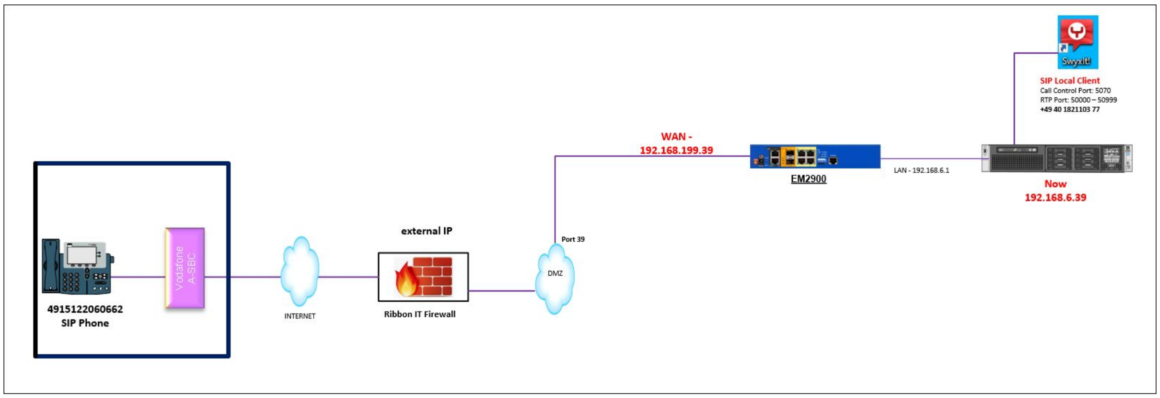
SIP Trunk Deployment Topology

The deployment topology diagram is depicted below.



Interoperability Test Lab Topology (or Call Flow Diagram)

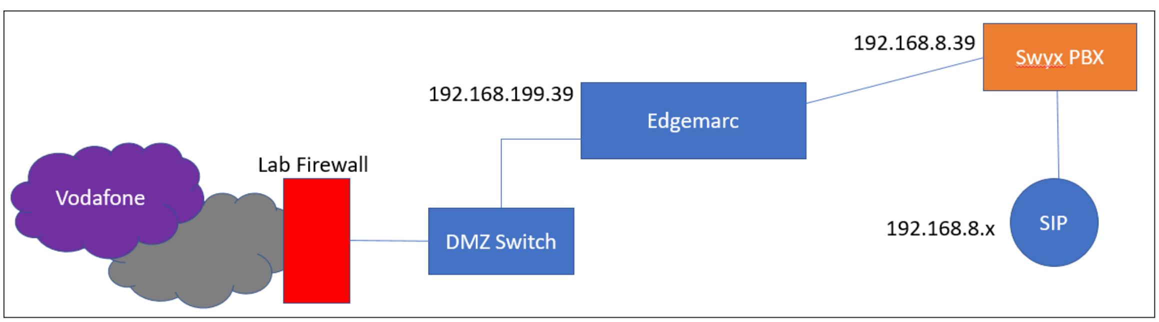
IOT high-level architecture covering call flows and overall topology is depicted below.



Section-A: EdgeMarc 2900 Configuration

Connectivity

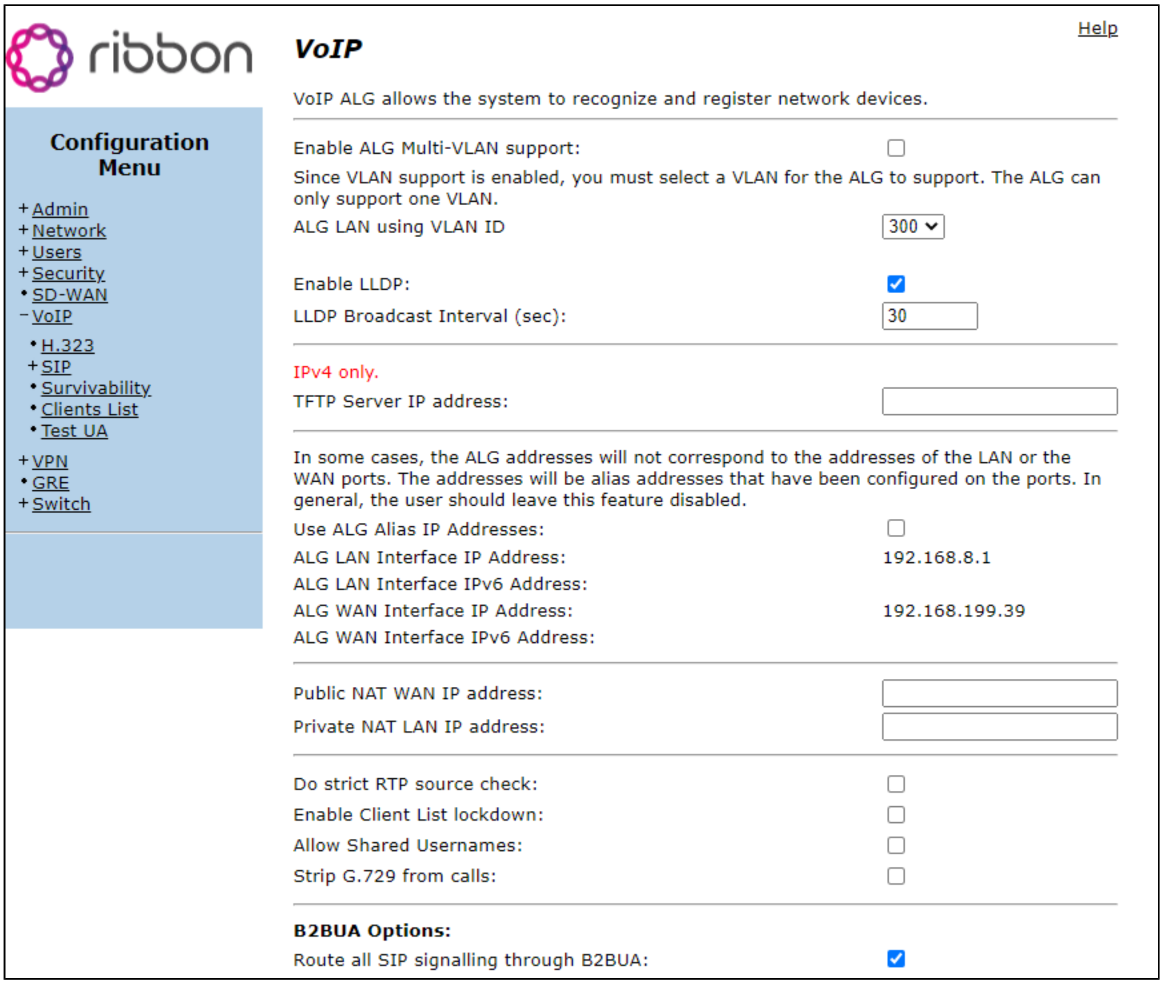
In this case, a Sonus 1000 is used as a stand-in for the SIP Service Provider. Call testing was performed from a SIP client registered to the Sonus 1000 to the HFA phone behind the EdgeMarc 2900 as well as to a Swyx soft client. The EdgeMarc is configured to apply its SIP ALG function to traffic between the Sonus 1000 and the Swyx PBX. This traffic is standard SIP on UDP port 5060. In this use case, the EdgeMarc does not need to handle any HFA protocol traffic. The voice configuration of the EdgeMarc represents a typical configuration for the EdgeMarc with few notes required.



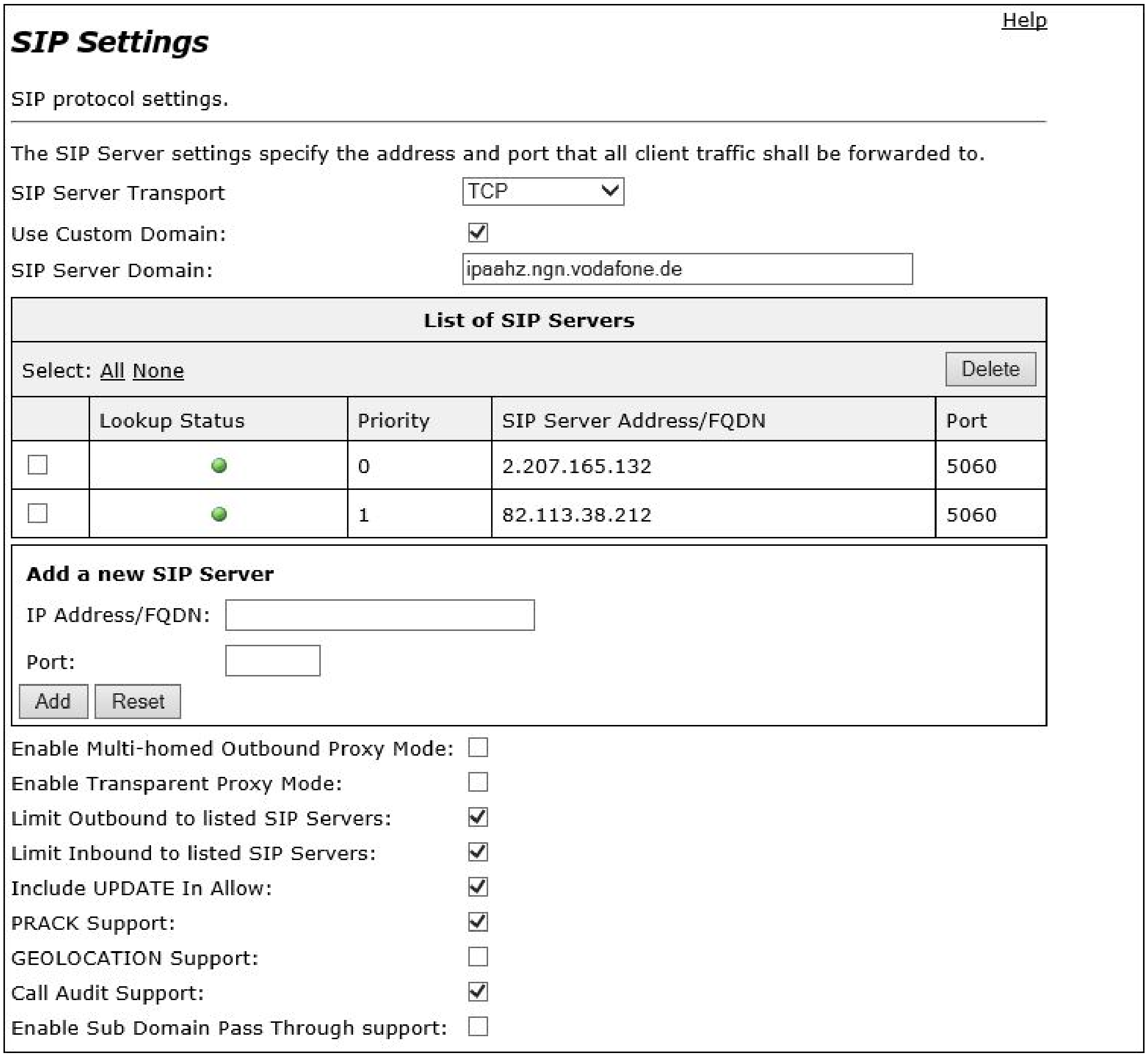
Configuring EdgeMarc ALG

1. The SIP ALG of the EdgeMarc is configured as follows:
   - In the ALG VLAN drop-down box, select the VLAN where the PBX resides.

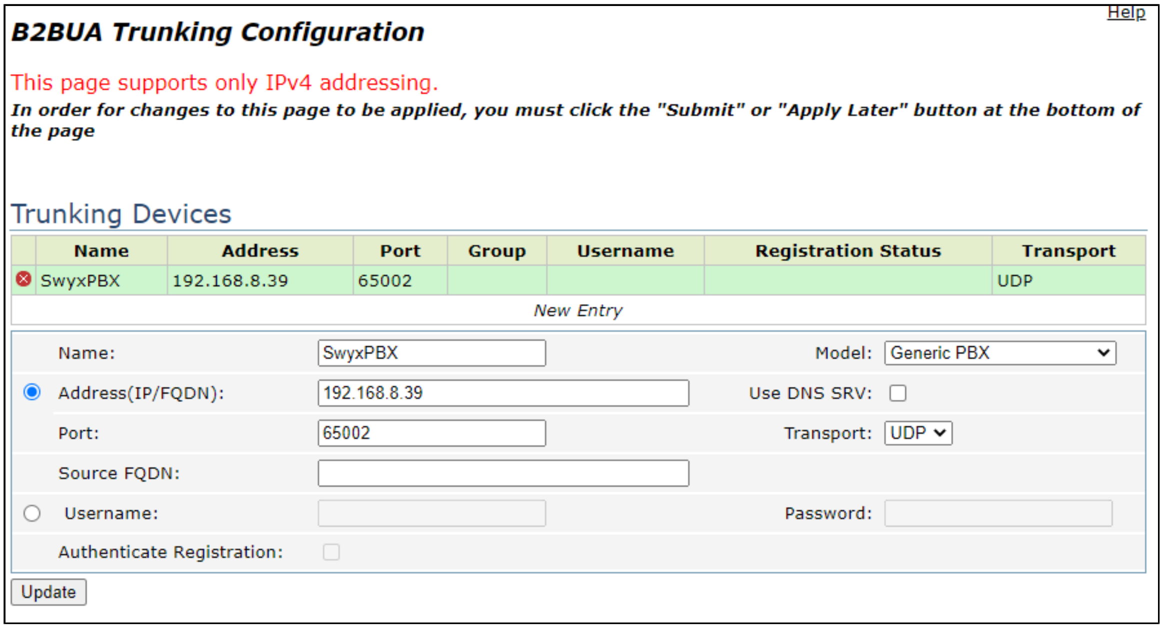
- Select Route all signaling through B2BUA to facilitate any SIP header manipulation requirements.



2. On the SIP Settings page, configure the WAN-side SIP. The SIP server address configured is that of the SIP Provider's SBC. In this case, it is the public IP of the Sonus 1000 in use for the testing. All other settings on the page are the default settings for the EdgeMarc.



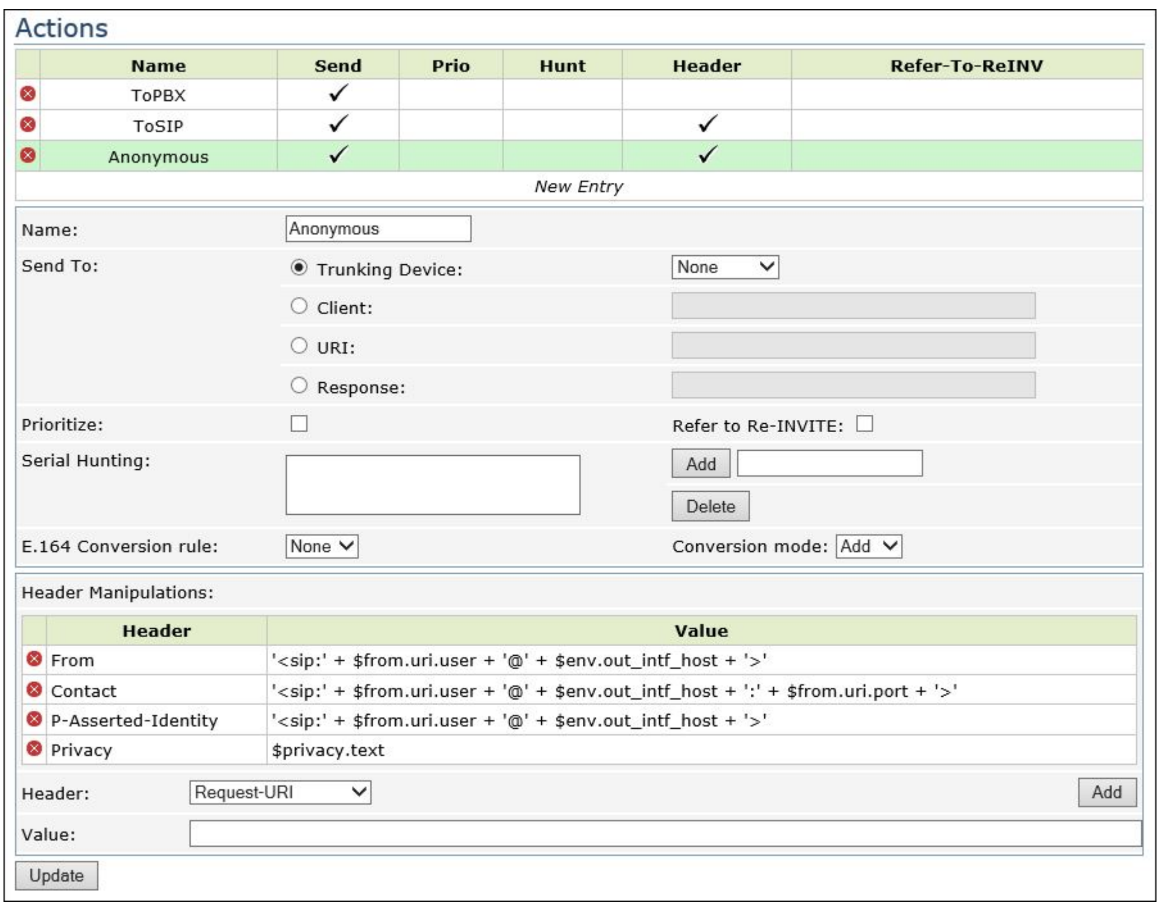
3. The LAN settings are found under VoIP SIP B2BUA. This is a multi-part configuration.
   Starting with the PBX SIP trunking configuration:



This creates a LAN-side destination to send and receive traffic from the SIP service provider.

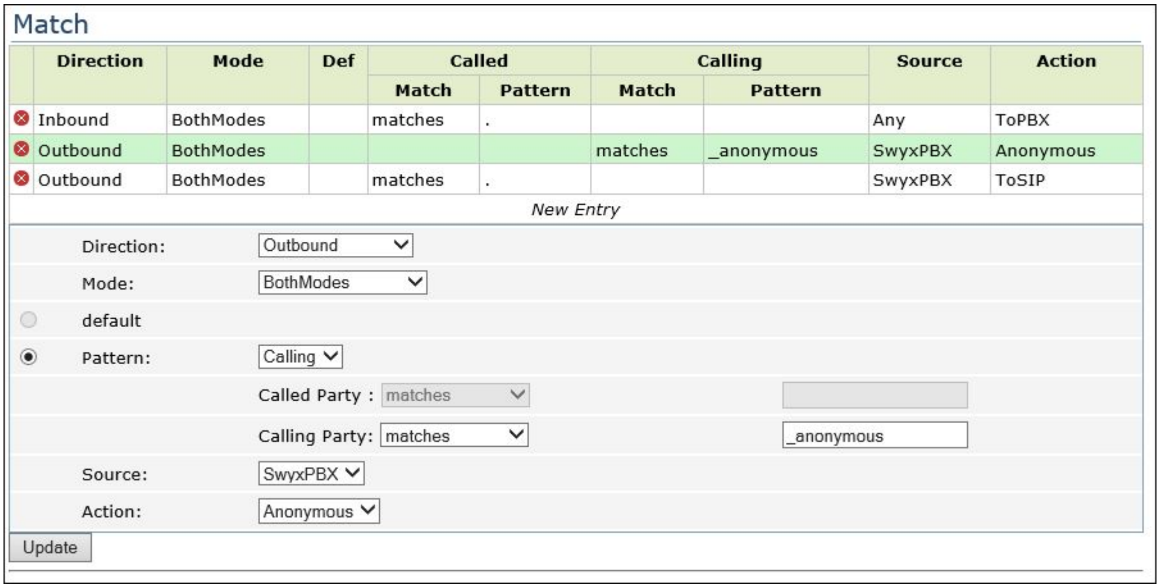
Note: The port is set to 65002 to accommodate a requirement in the Swyx PBX.

4. Configure the Actions to route calls in the proper direction:
   - The ToPBX routes calls to the trunking destination of SwyxPBX. This is the entity created in the previous step.
   - The ToSIP action is a default action to route specific calls towards the Service Provider. Although no header manipulation rules were required or configured for testing, they may be added here if needed (for example, to support digit manipulation or other accommodations for the service provider). Refer to EdgeMarc VOS documentation for syntax and capabilities.
     Note: Action "Anonymous" was added so that CLIR information would be passed on from the LAN side to the WAN side.



5. Finally, configure call routing in the Match section. A basic dial plan is used in the lab that routed all calls to the Swyx PBX with the exception of a single endpoint registered to the Sonus 1000 acting as the SIP provider.

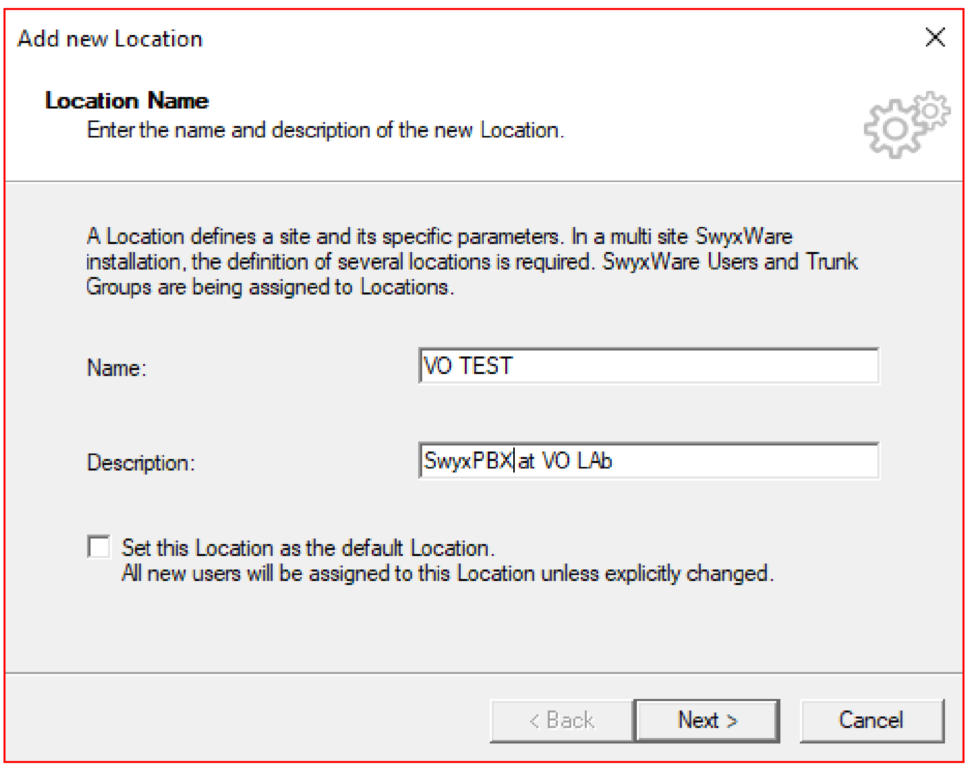
Note: Match for Action "Anonymous" was added so that CLIR information would be passed on from the LAN side to the WAN side.



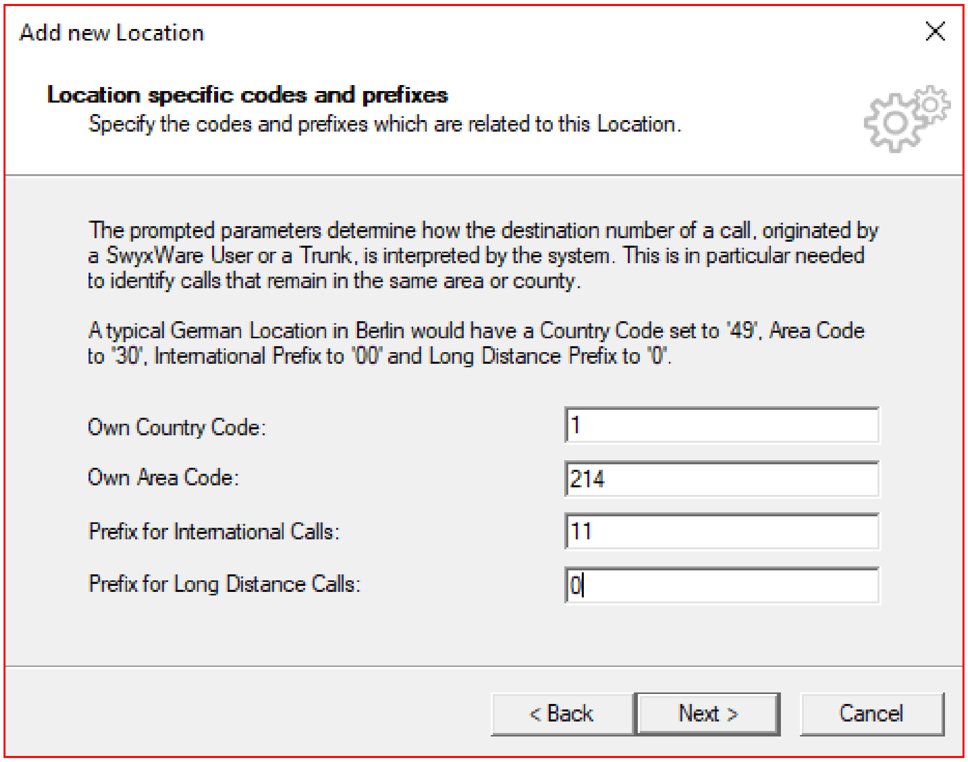
Section-B: SwyxWare, SwyxIt, and HFA Phone Configuration

Configuring SwyxPBX

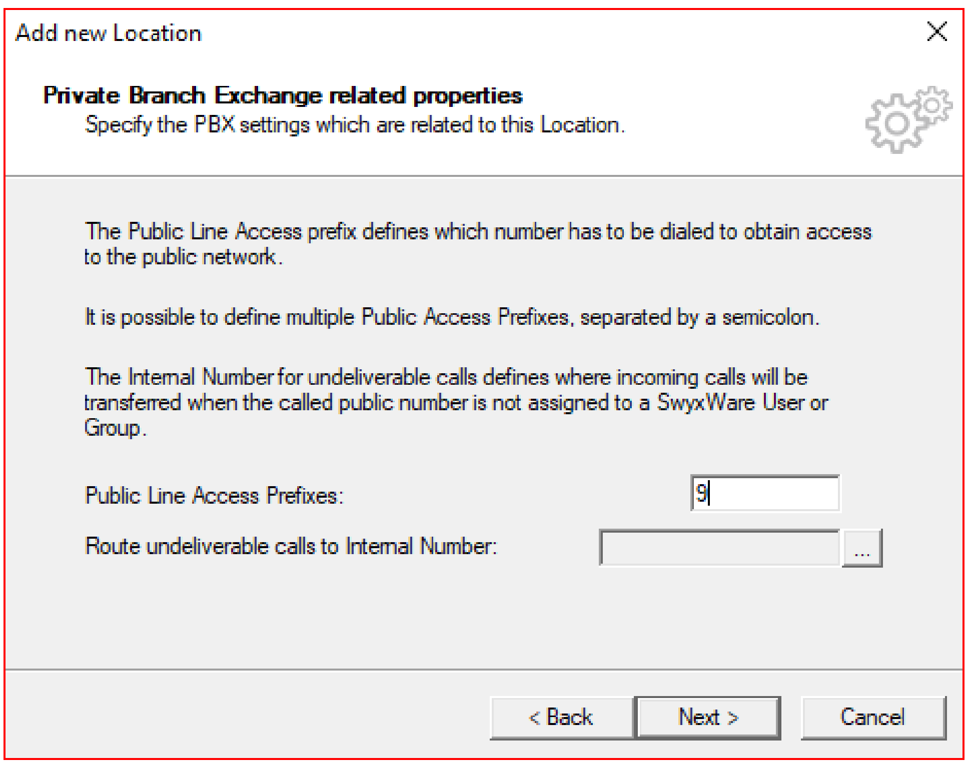
1. Right-click on Location > Add Location.



2. Add codes and prefixes, and click Next.

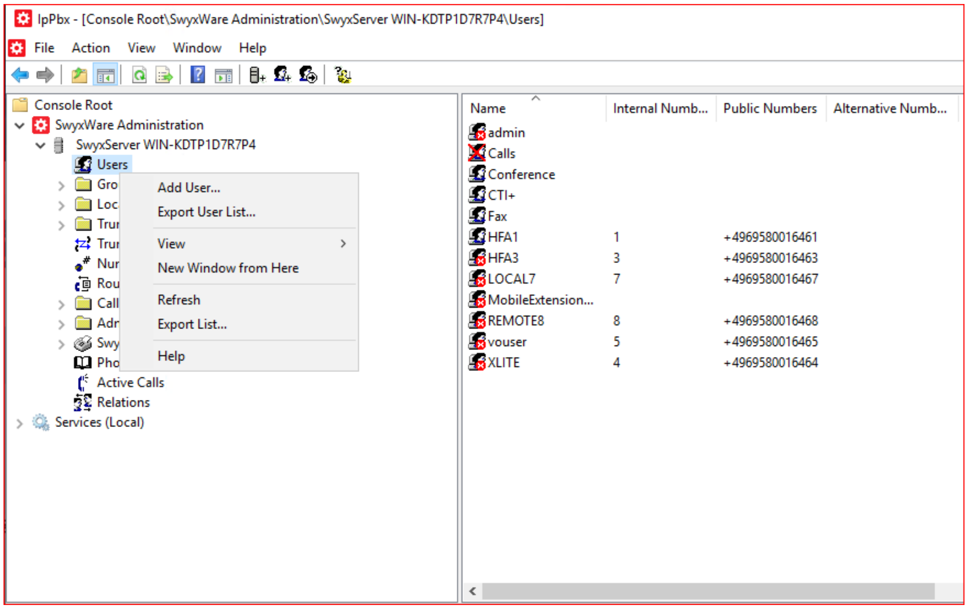


3. Add access to Dial out and click Next.

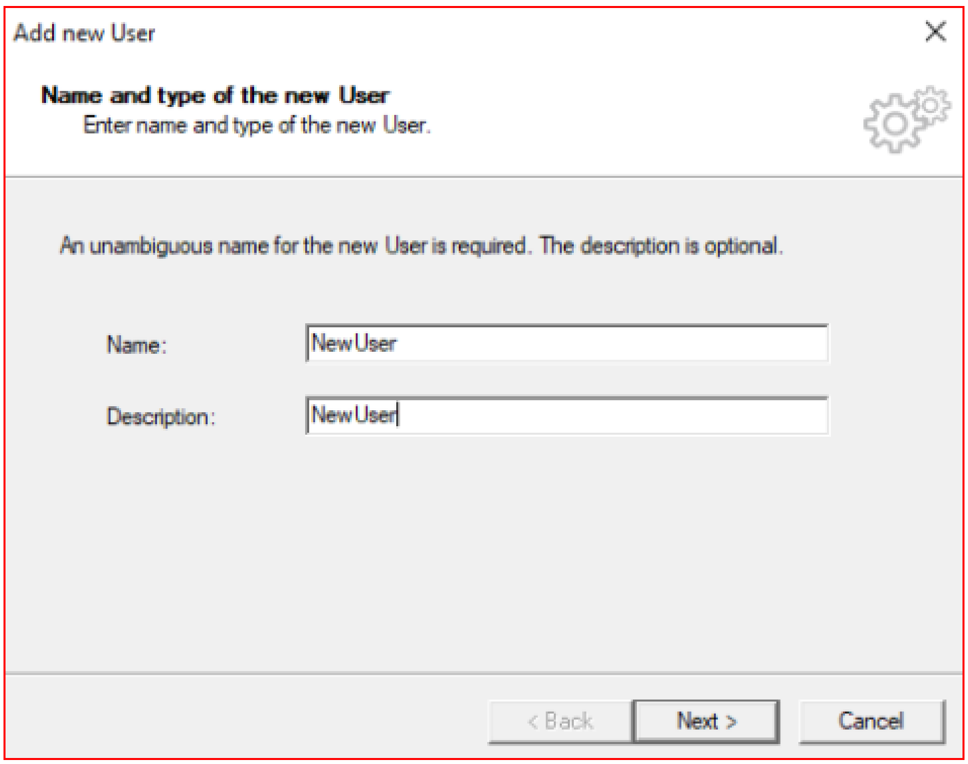


Adding a User

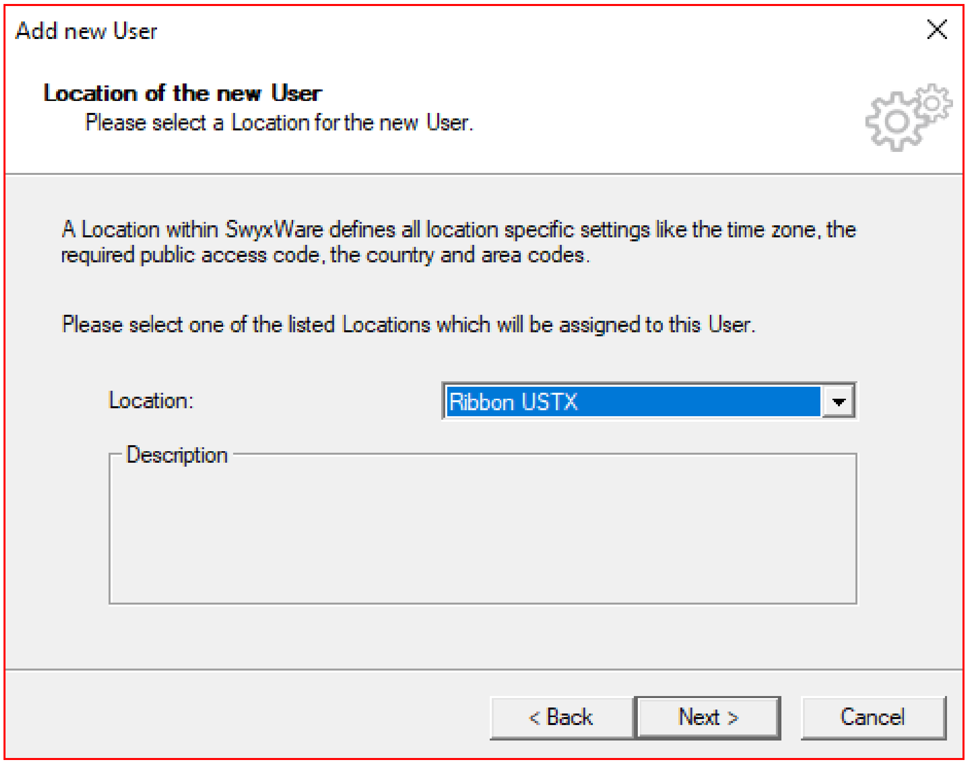
1. Click on Server > right-click on User > click Add User.



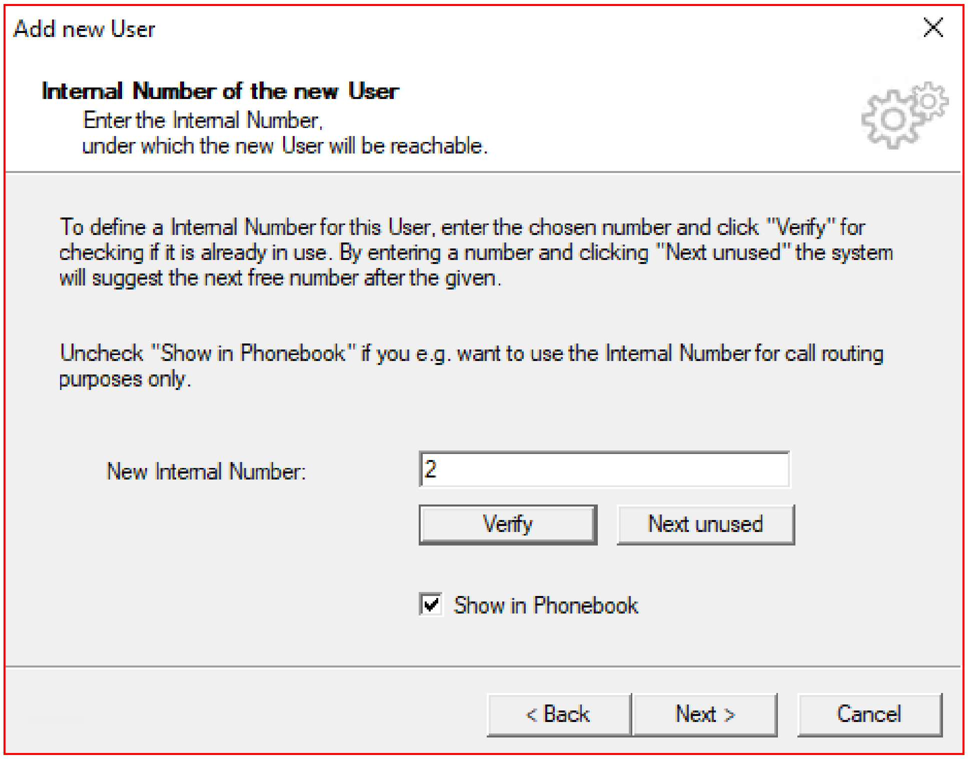
2. Add Name and Description, then click Next.

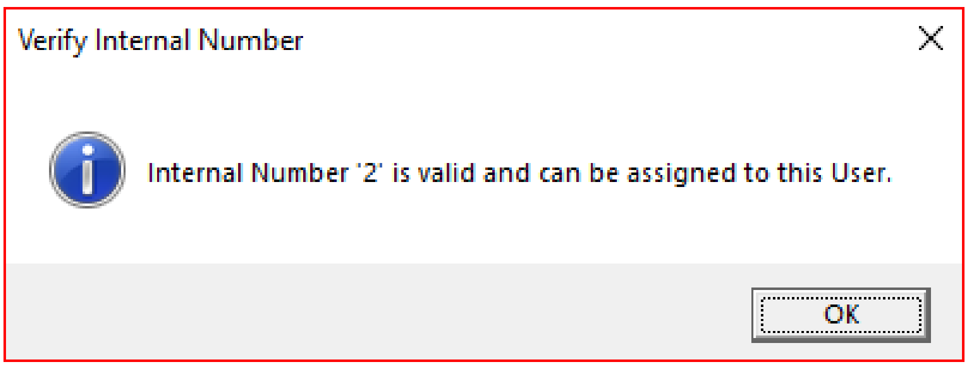


3. Select the Location and click Next.

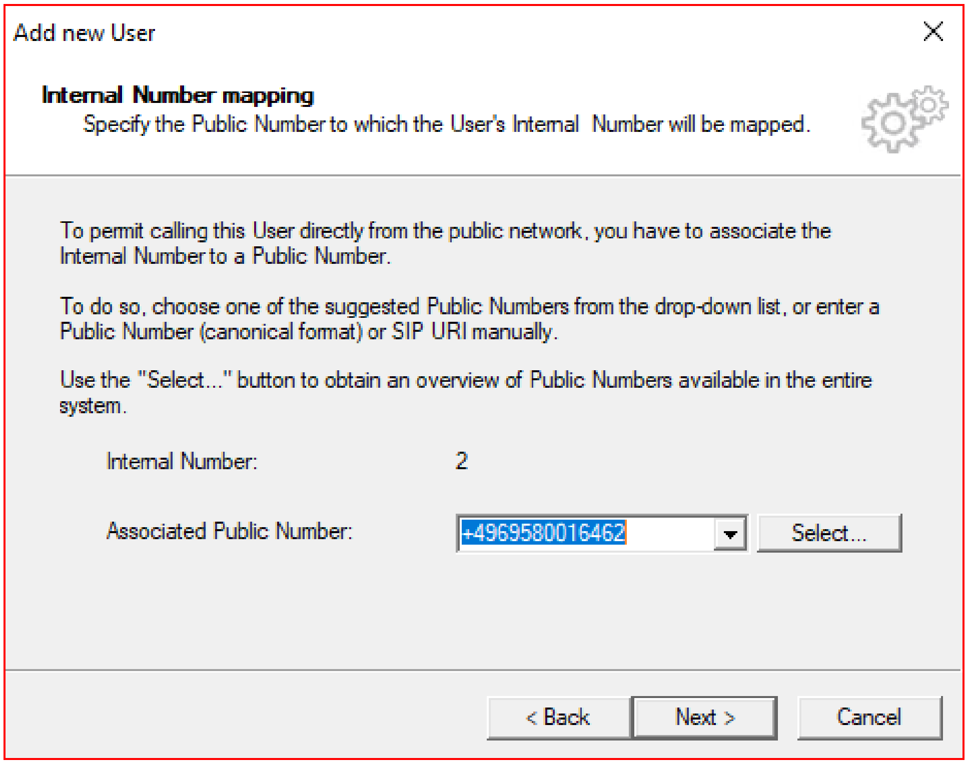


4. Select a new internal number and click Verify to check if it is available. Click OK, then click Next.

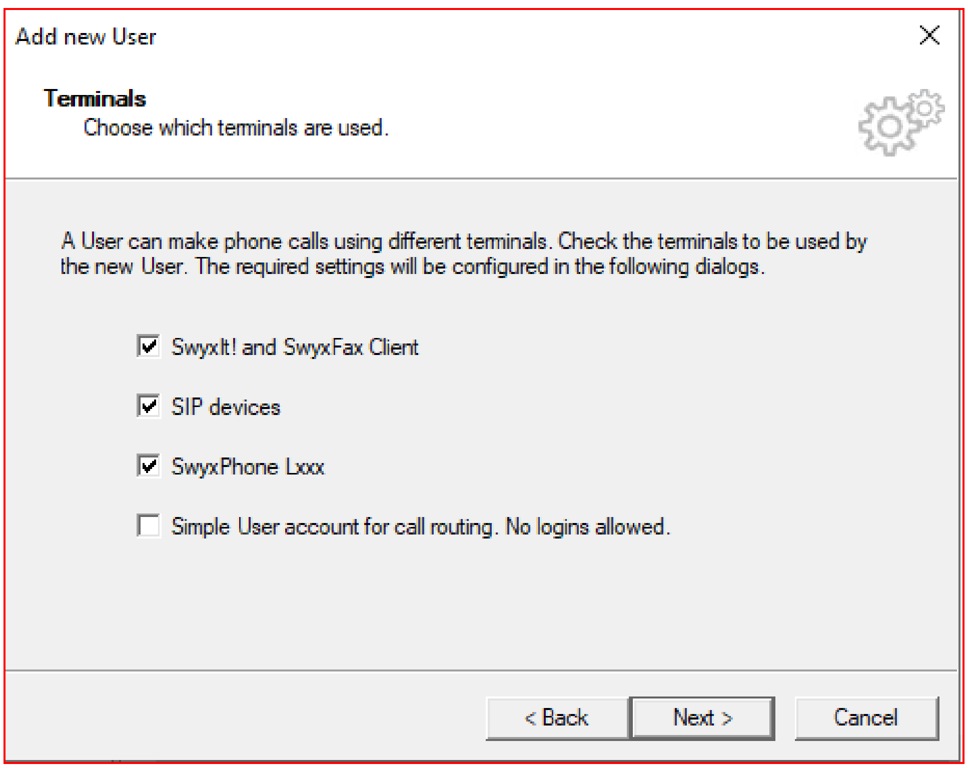

Add new User
Internal Number of the new User
Enter the Internal Number,
under which the new User will be reachable.
To define a Internal Number for this User, enter the chosen number and click "Verify" for checking if it is already in use. By entering a number and clicking "Next unused" the system will 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
Verify
Next unused
Show in Phonebook
< Back
Next >
Cancel


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

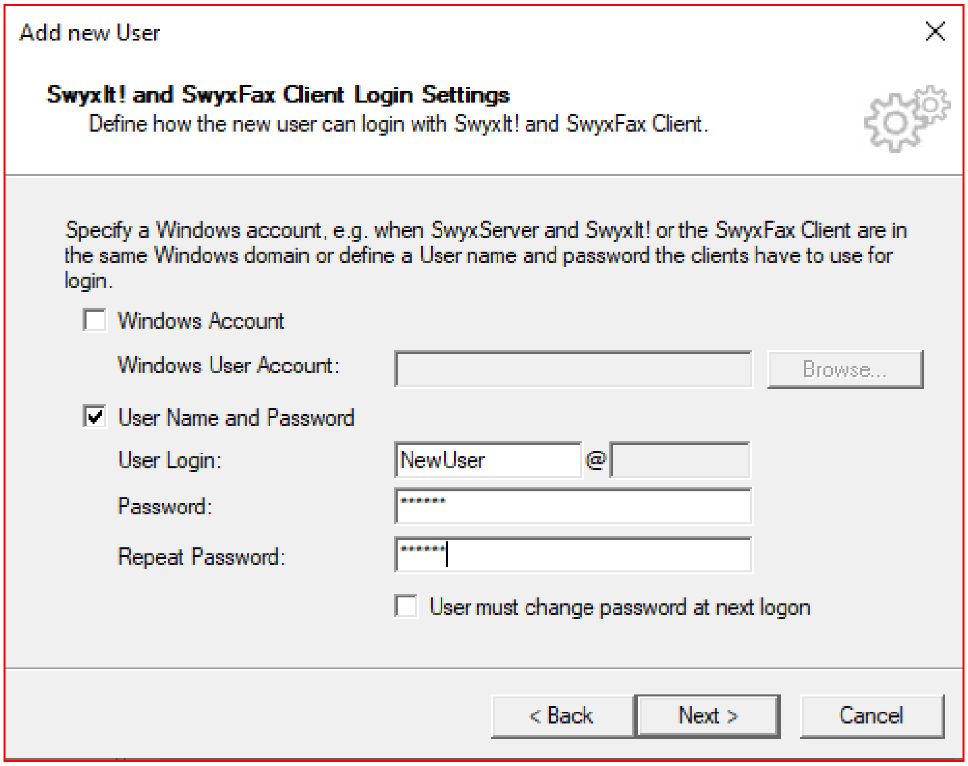
5. The Internal Number selected will be mapped to a public number, then click Next.



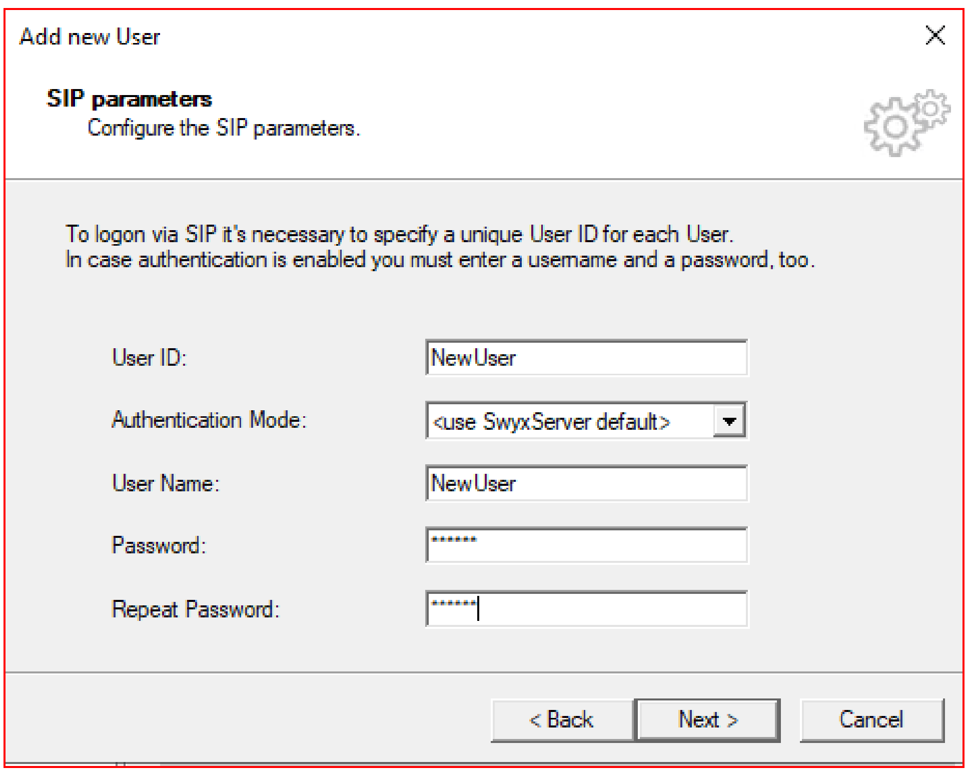
6. Select the Terminals by checking the boxes and click Next.



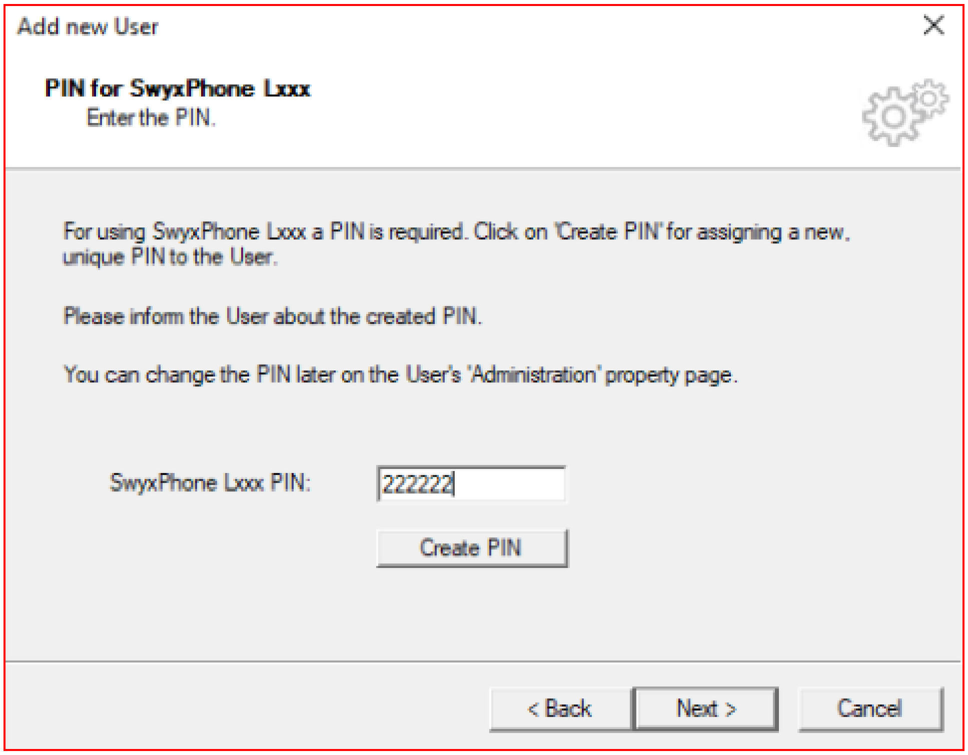
7. Create a Password for User Login, then click Next.



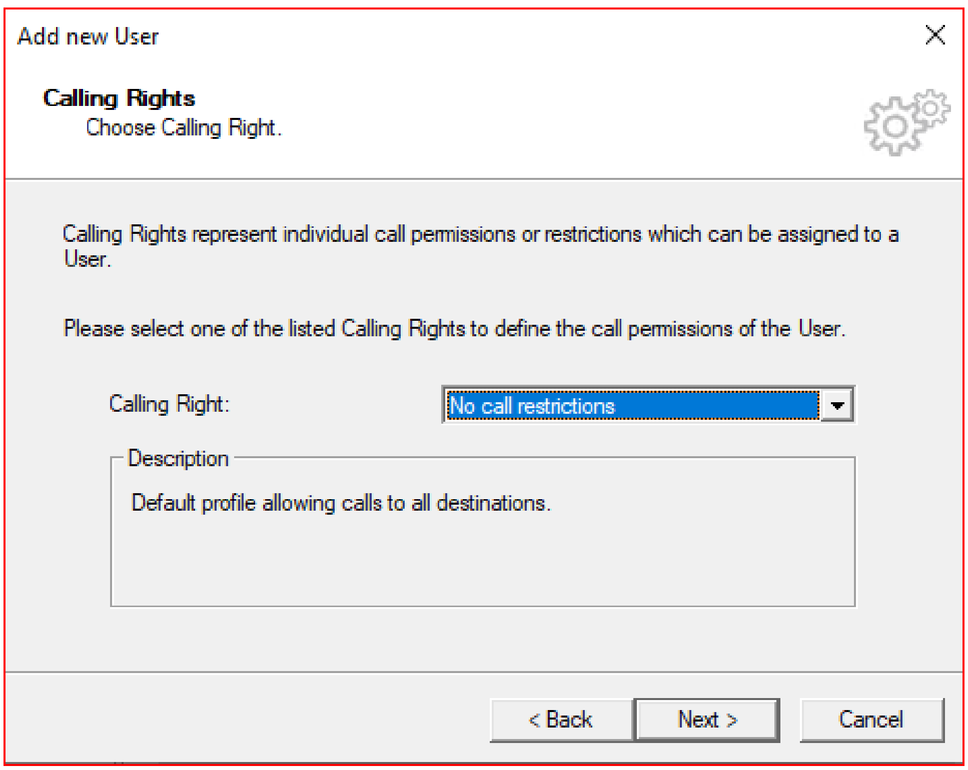
8. Create a SIP user and password, then click Next.



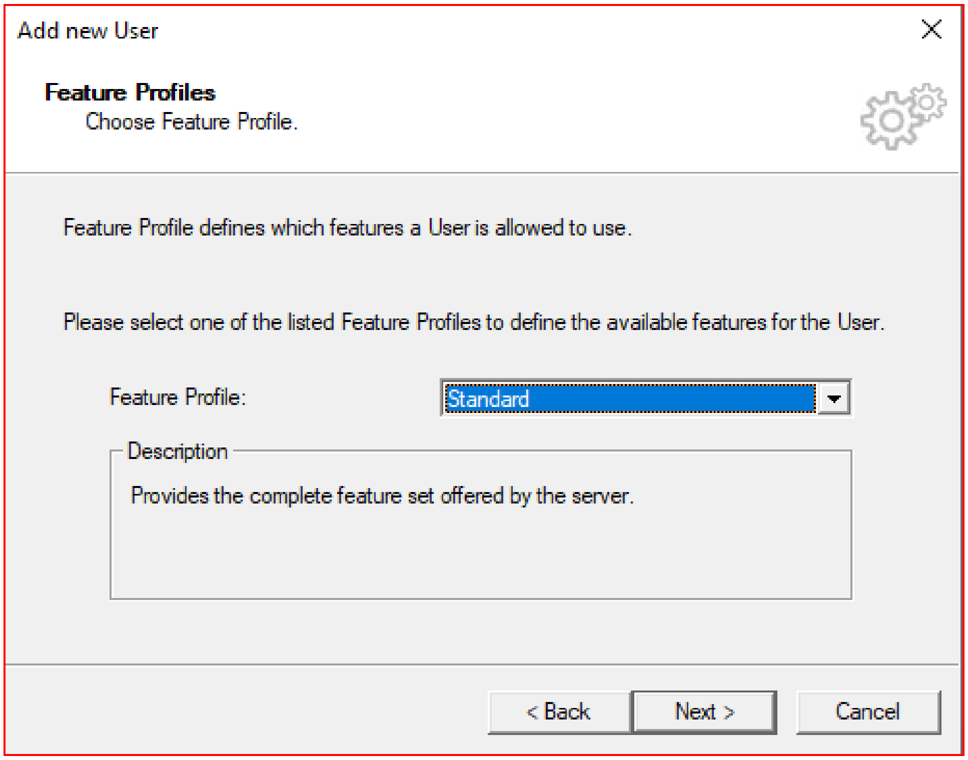
9. Create a Swyx Phone Pin, then click Next.



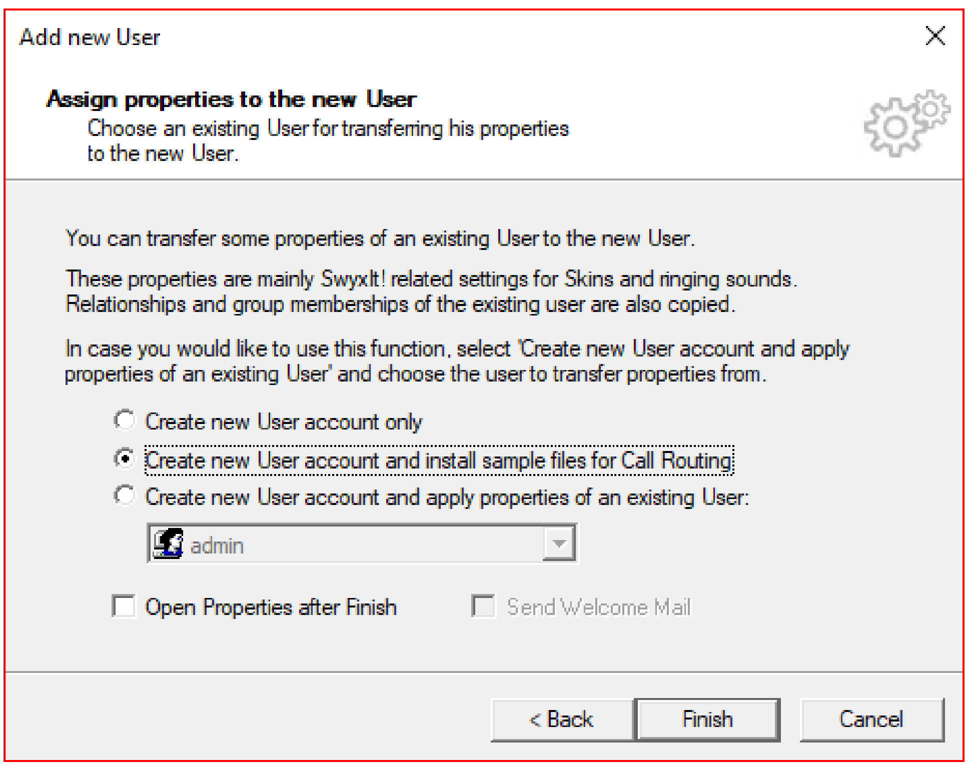
10. Select a Calling Right, then click Next.



11. Select a Feature Profile and click Next.

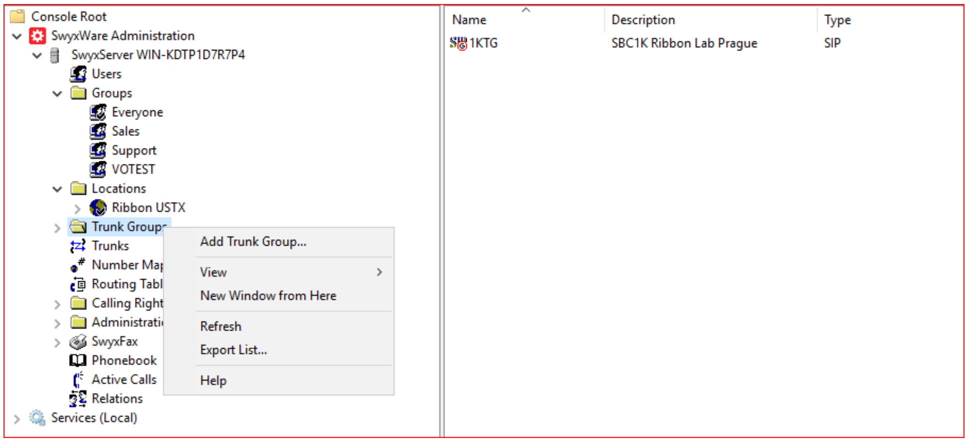


12. Assign Properties to the new user and click Finish.

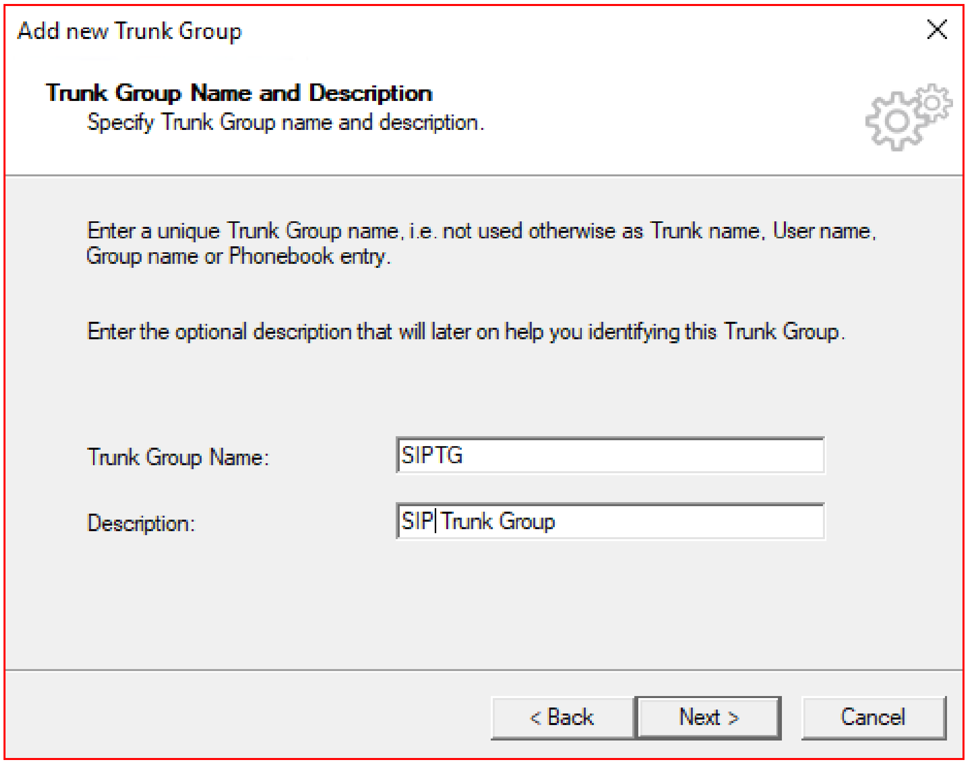


Configuring a SIP Trunk

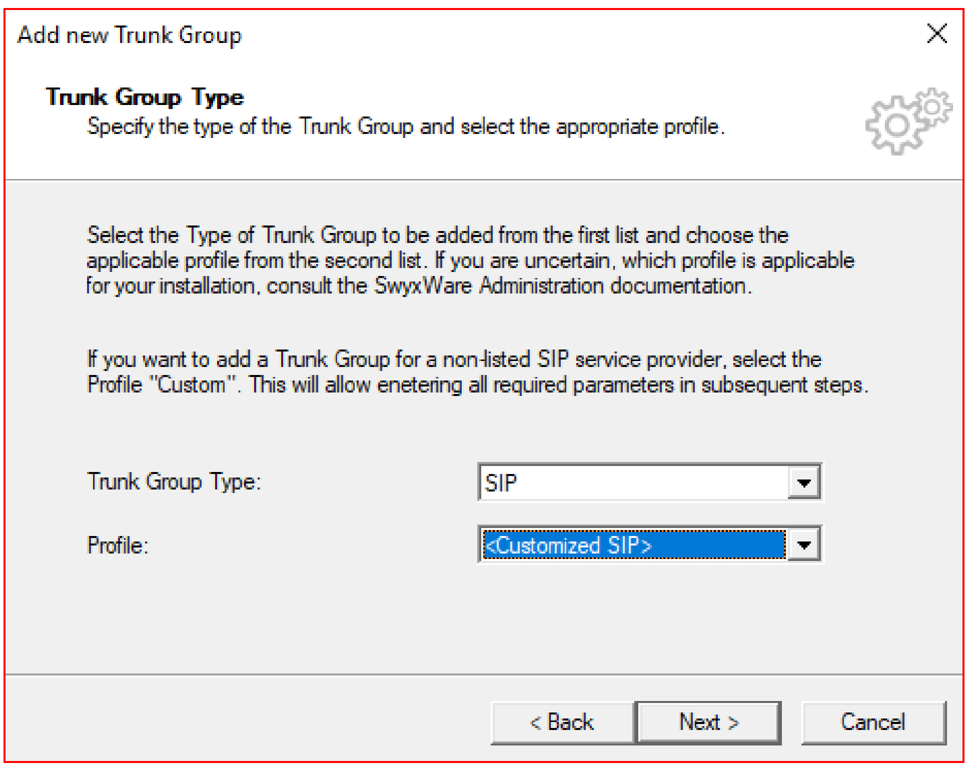
1. Right-click on Trunk Group and select Add Trunk Group. The Add Trunk Group Wizard pops up, then click Next.



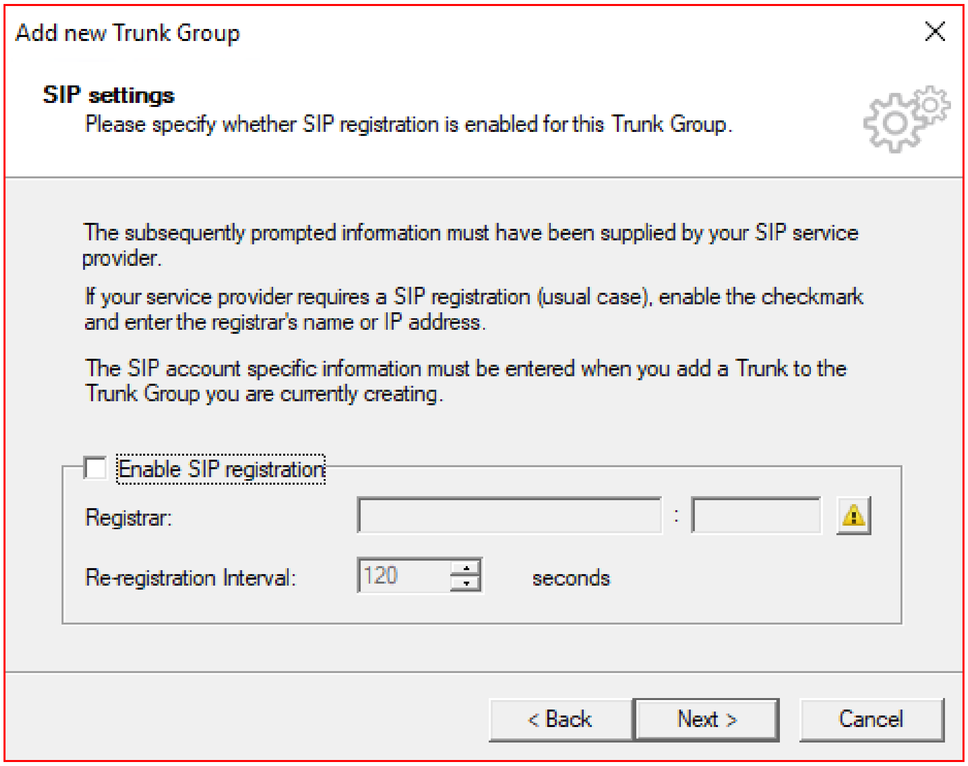
2. Add the Trunk Group Name and Description, and click Next.



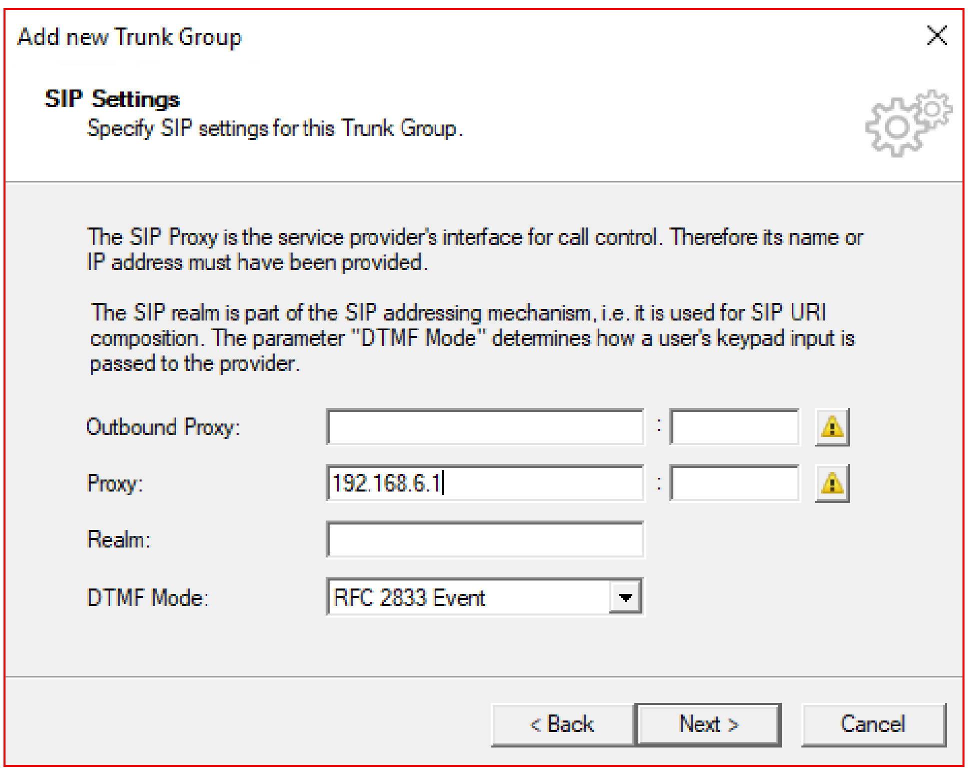
3. Select the Trunk Group Type and click Next.



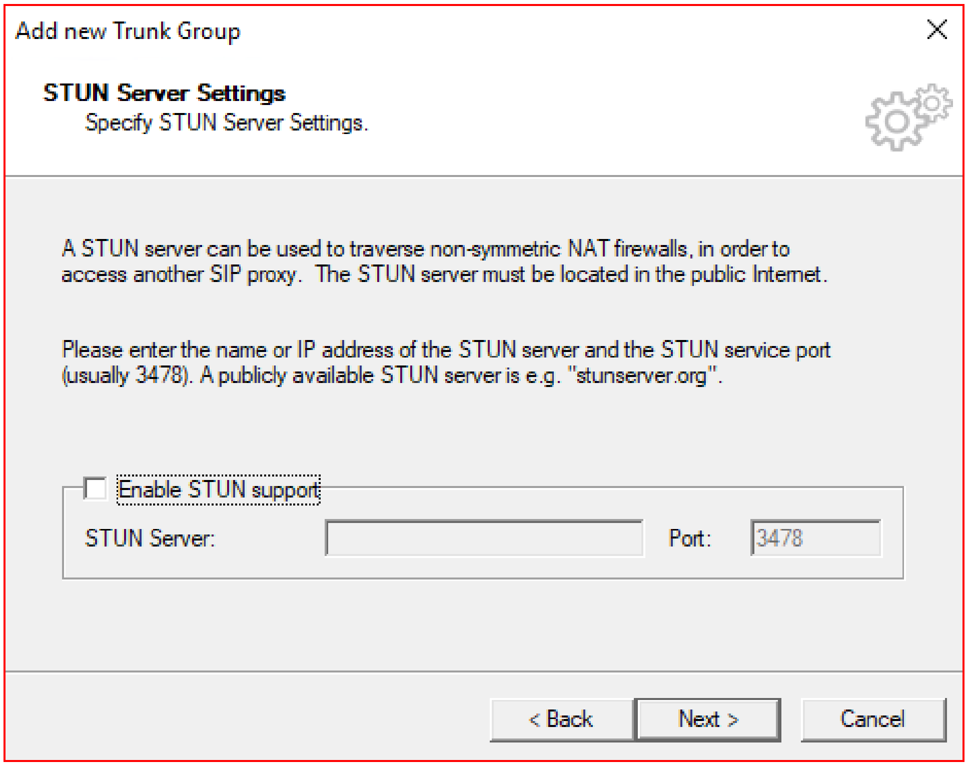
4. Add SIP settings and click Next.



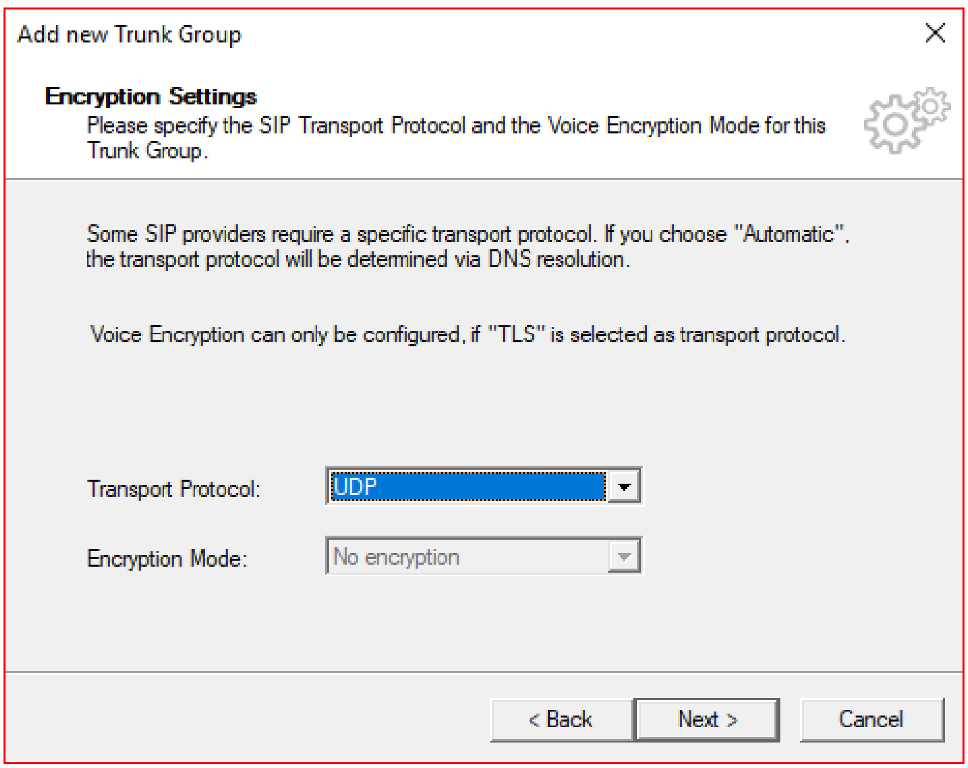
5. Add the SIP Proxy and leave SIP port blank (auto-resolves), and click Next.



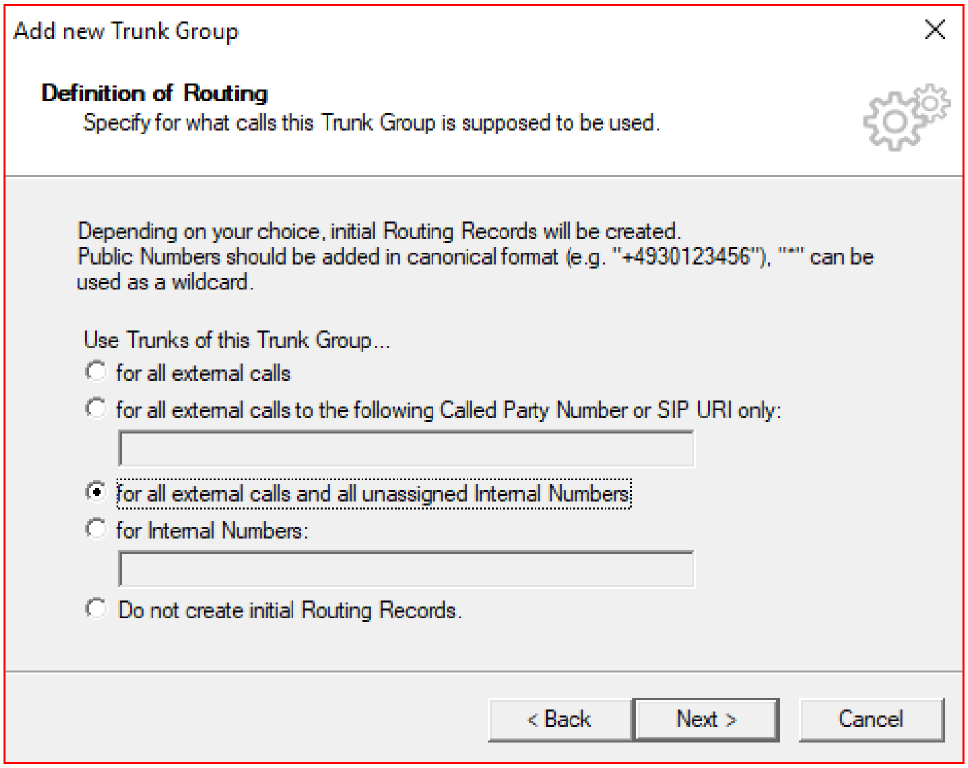
6. If a STUN server is supported, check the box and add the IP for STUN server, and click Next.



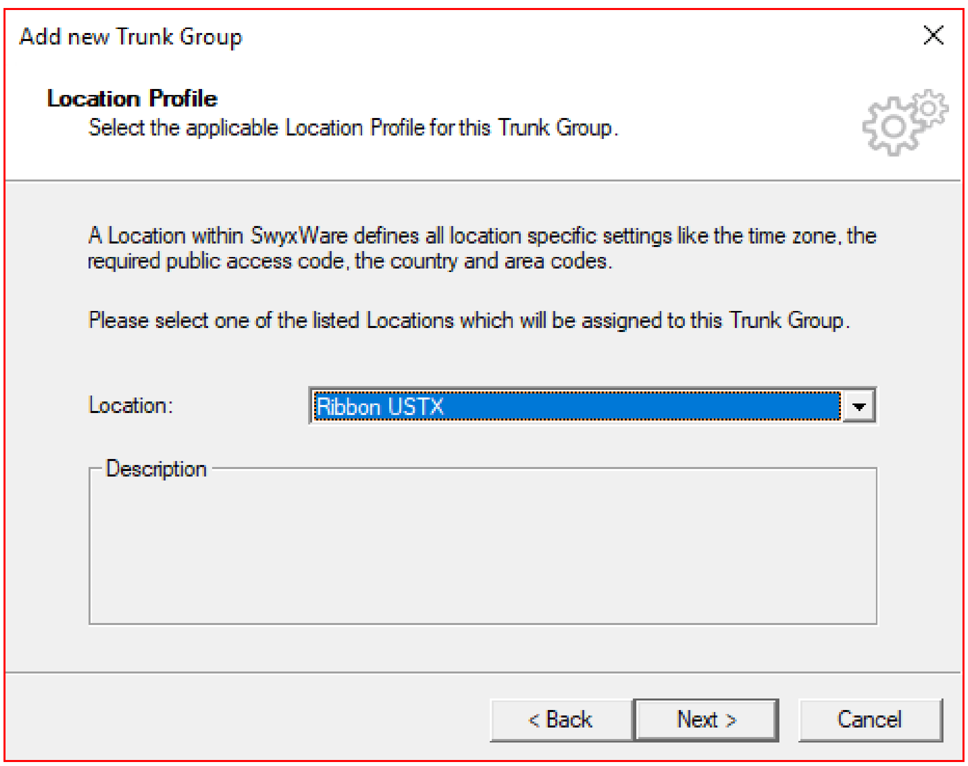
7. Select the Transport Protocol and click Next.



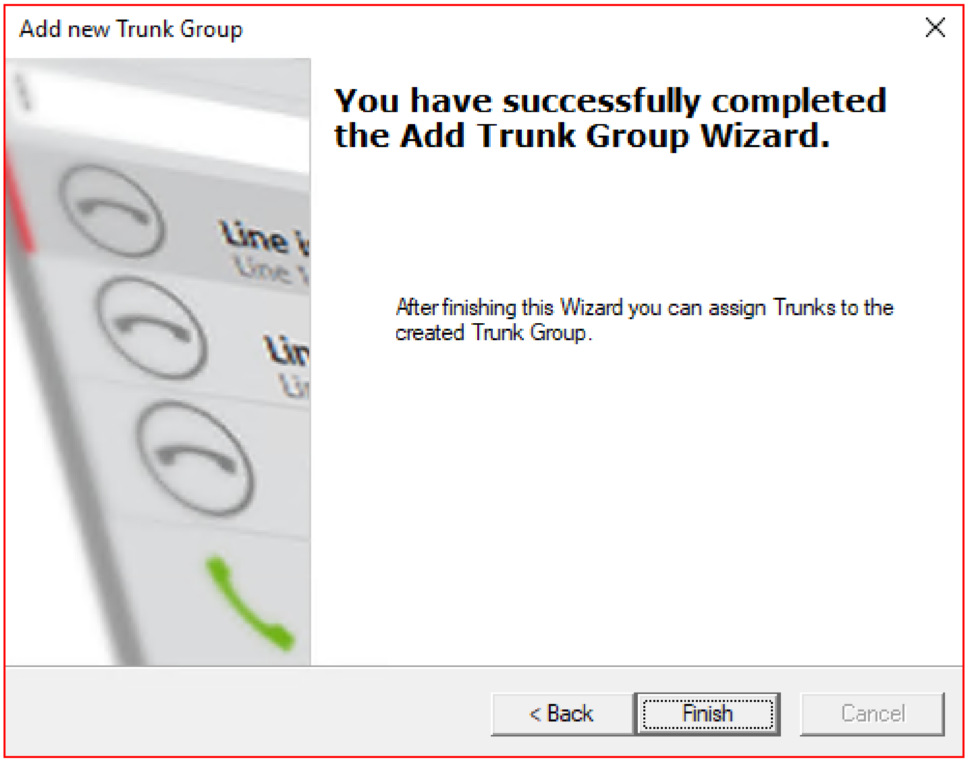
8. Select the Routing Definition and click Next.



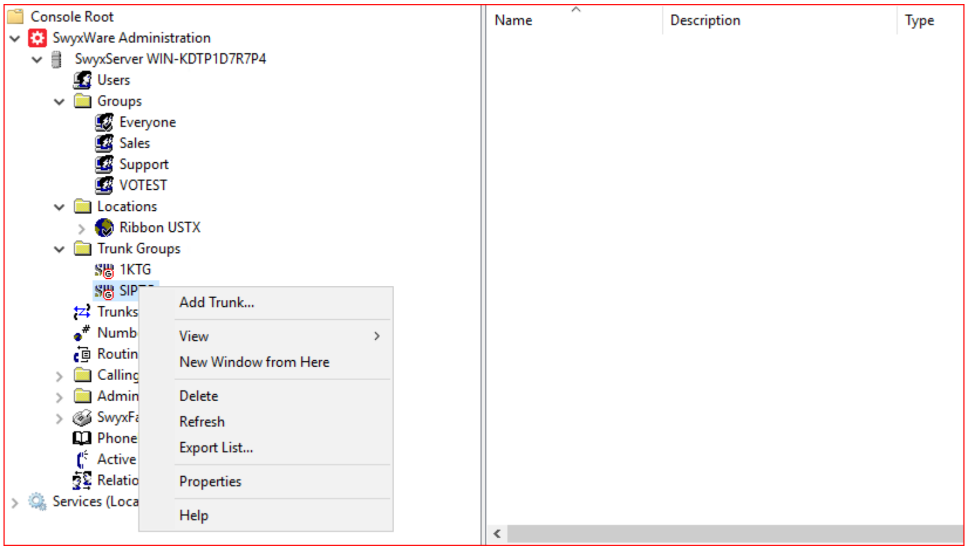
9. Select the Location Profile and click Next.



10. Click Finish.

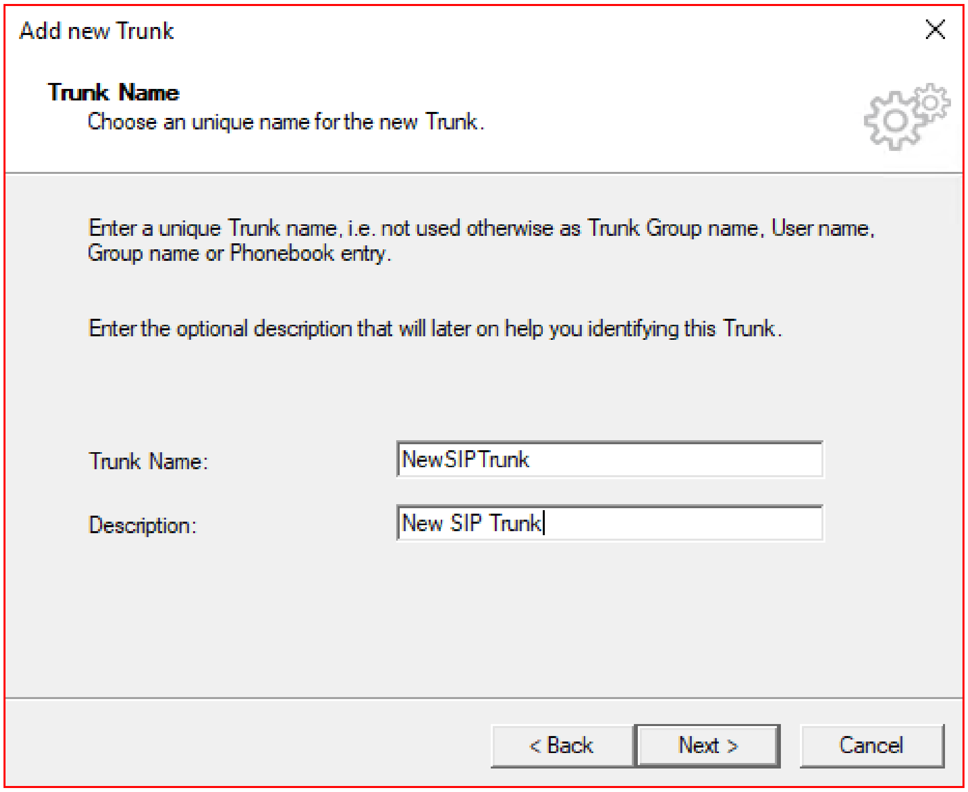


11. Right-click on the newly-created Trunk Group and Add Trunk. The Add Trunk Wizard pops up, then click Next.

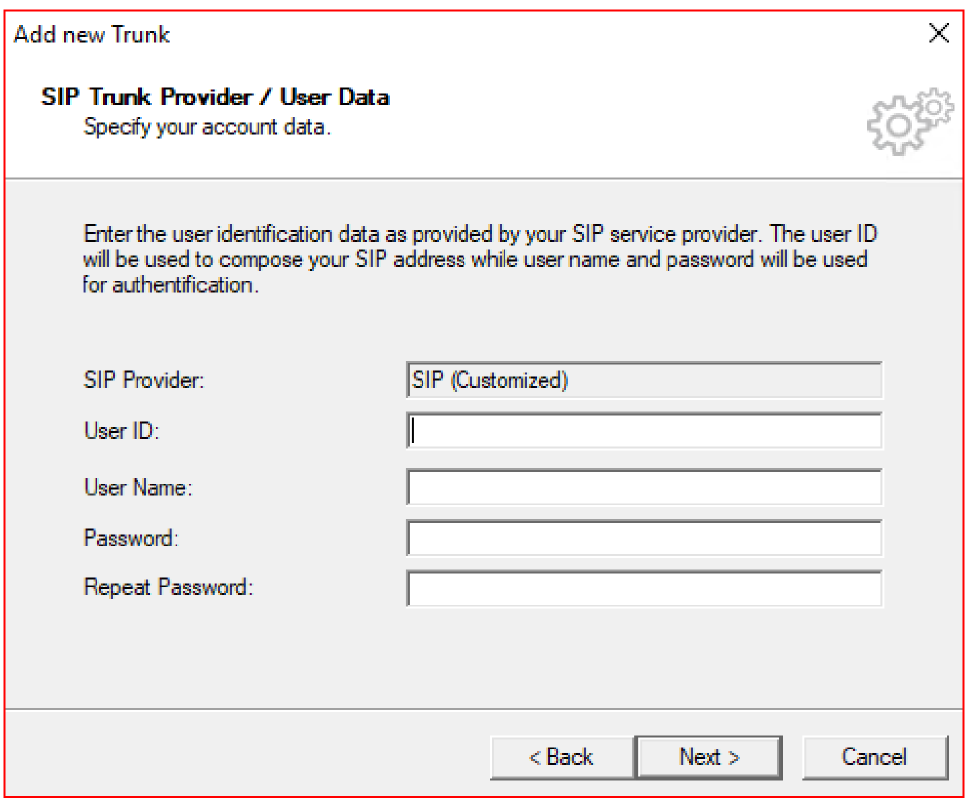

Console Root
SwyxWare Administration
SwyxServer WIN-KDTP1D7R7P4
Users
Groups
Everyone
Sales
Support
VOTEST
Locations
Ribbon USTX
Trunk Groups
1KTG
Add Trunk...
View
New Window from Here
Delete
Refresh
Export List...
Properties
Help
Name
Description
Type


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.
< Back
Next >
Cancel

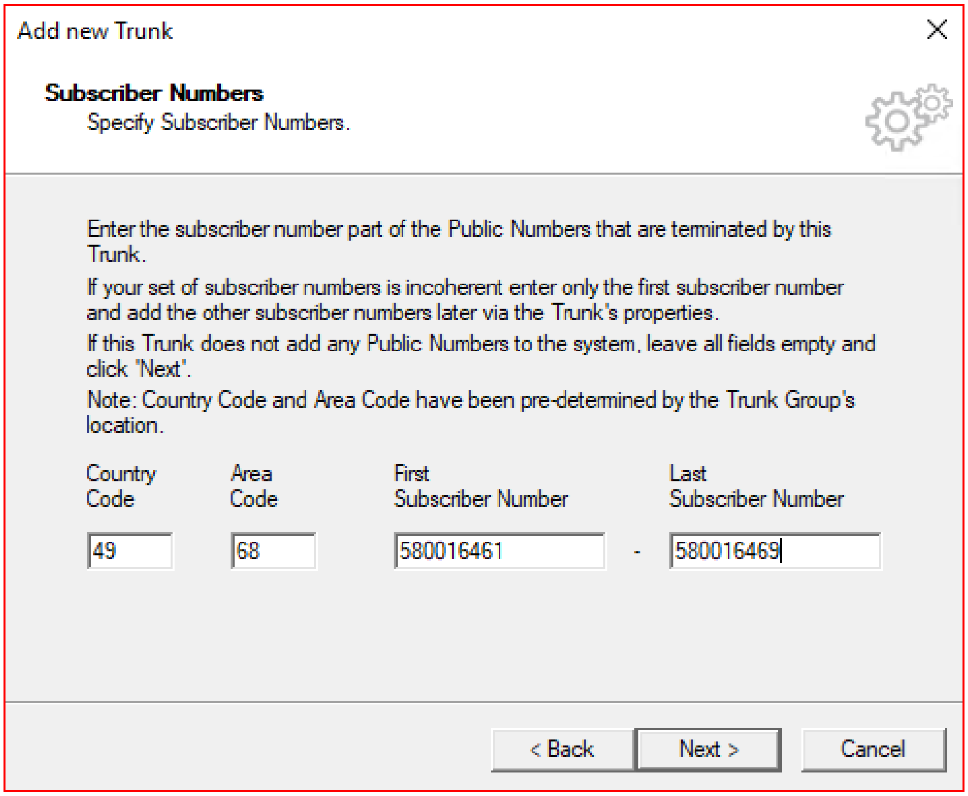
12. Add a Trunk Name and Description.



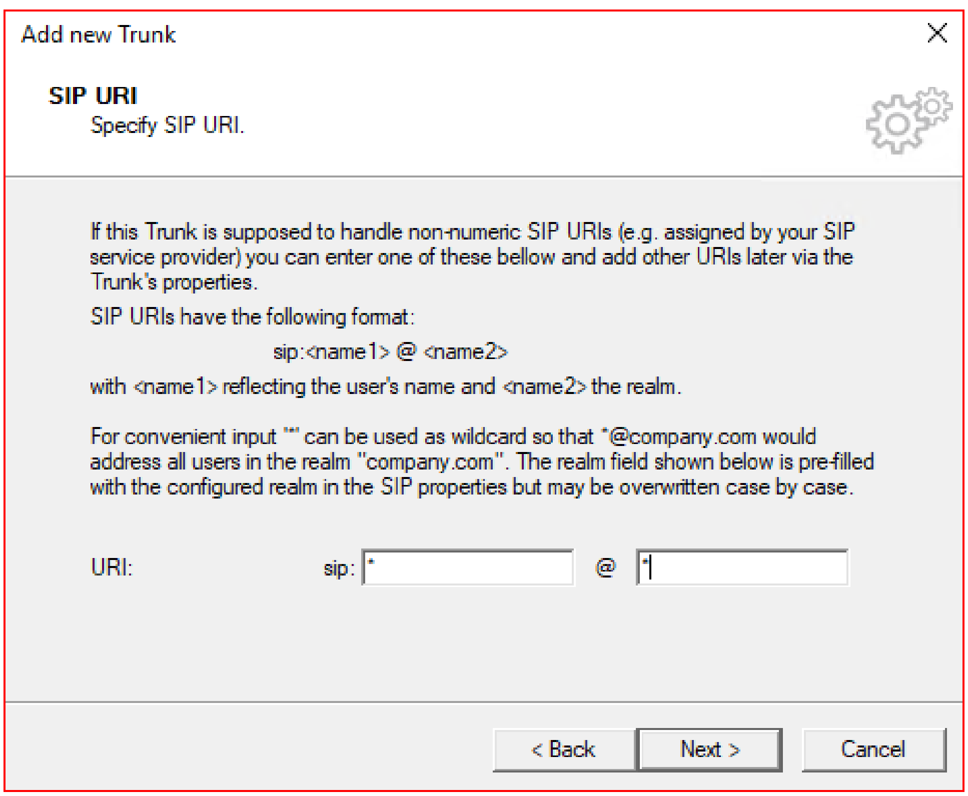
13. Add the SIP trunk Provider and User Data, and click Next.



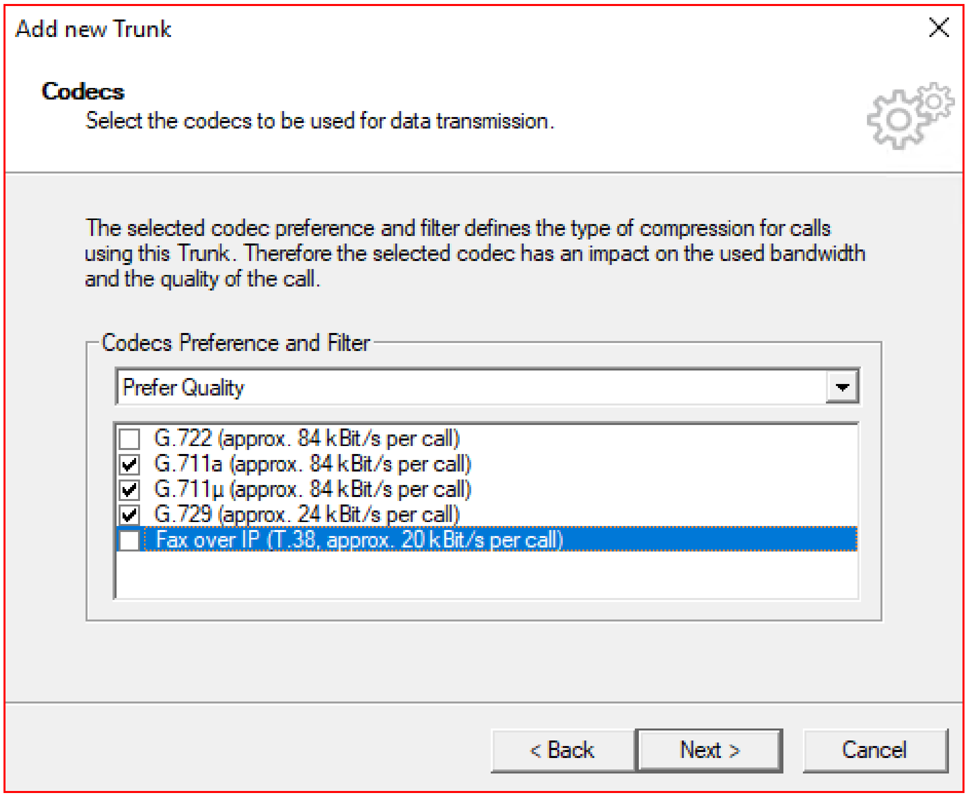
14. Select the Subscriber Number using the SIP Trunk and click Next.



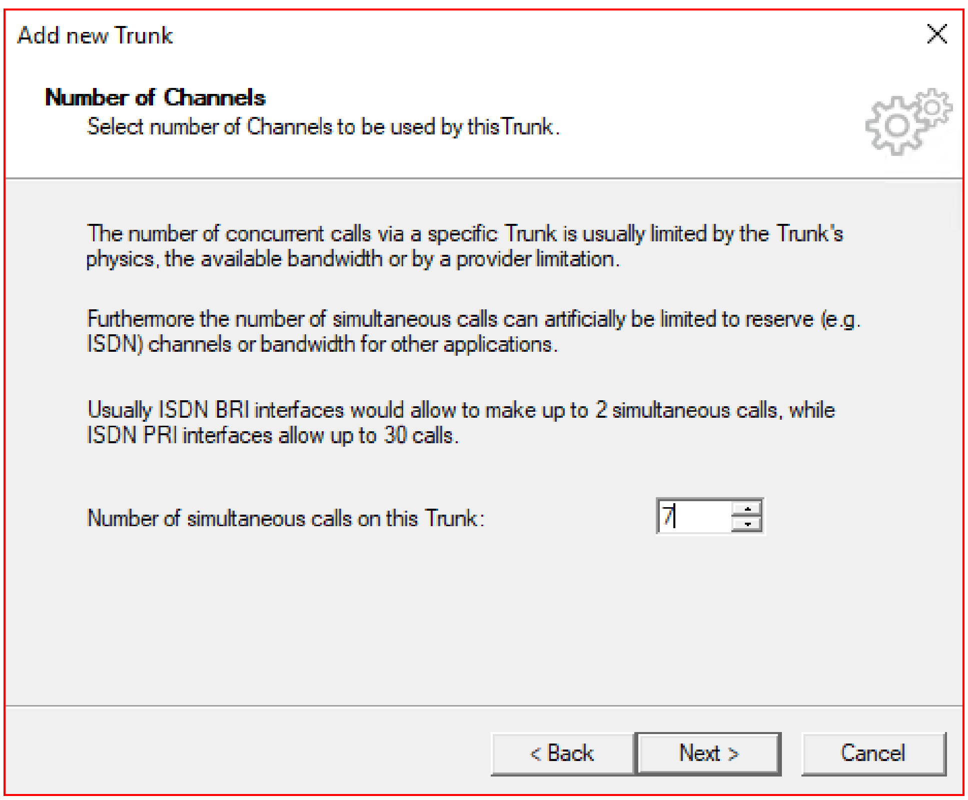
15. Add a SIP URI (wild card "*" for any), then click Next.



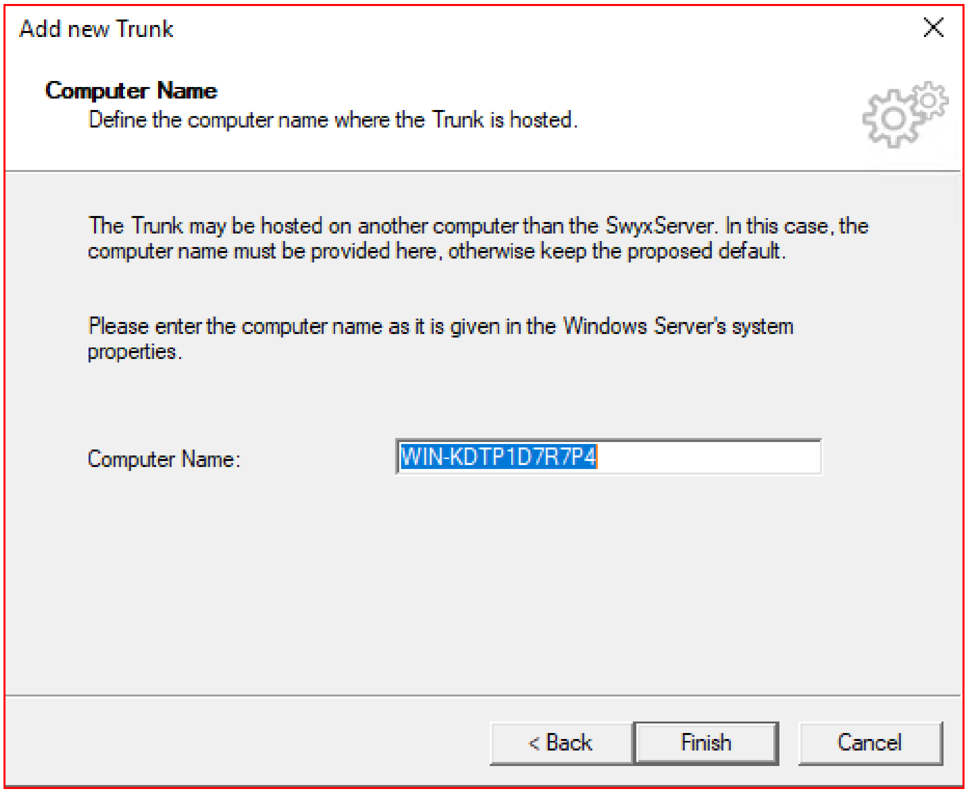
16. Select the Codecs supported by the SIP trunk and click Next.



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

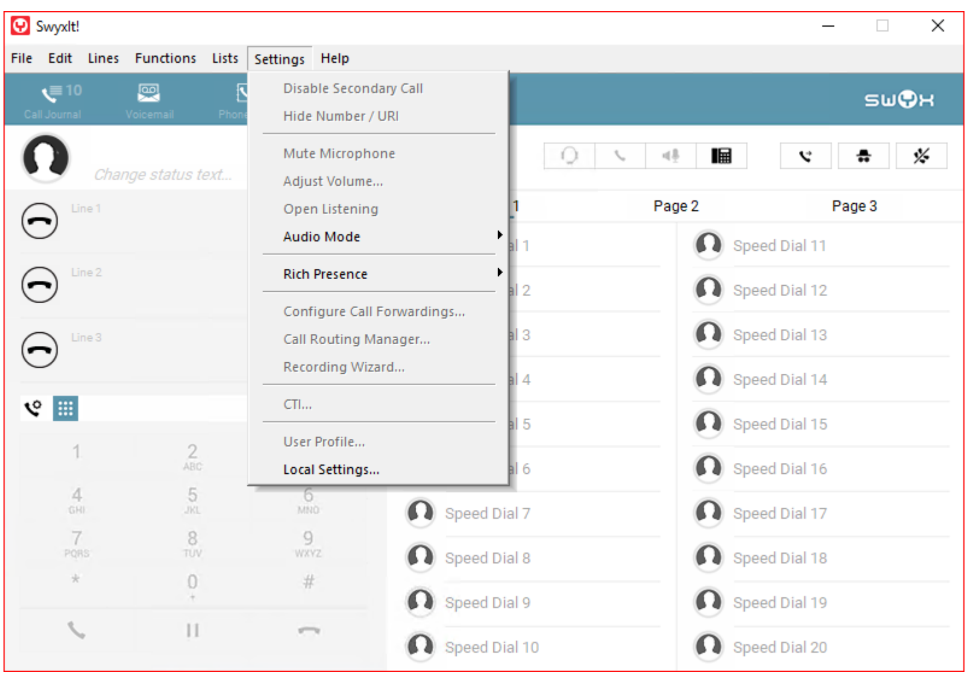


18. Choose PSX server or Computer Name and click Finish.

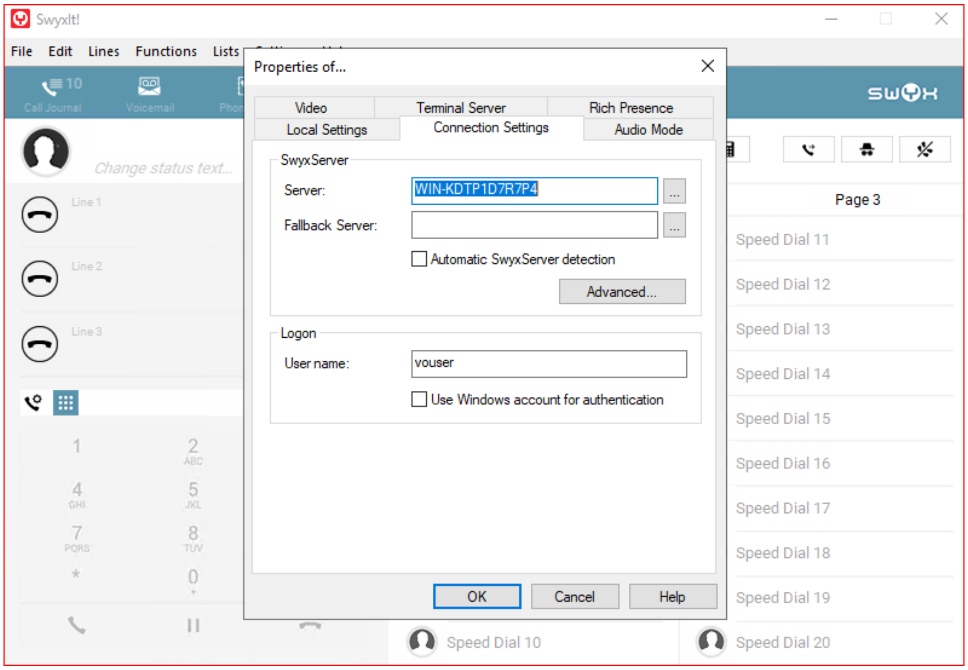


Configuring the SwyxIt Client

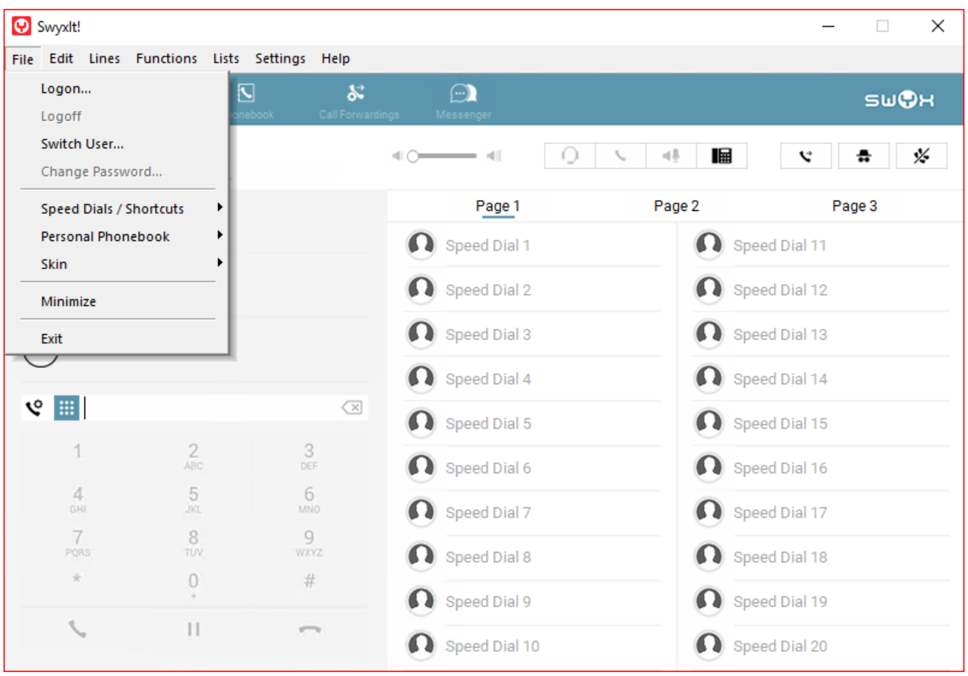
1. Click On Settings and select Local Settings.



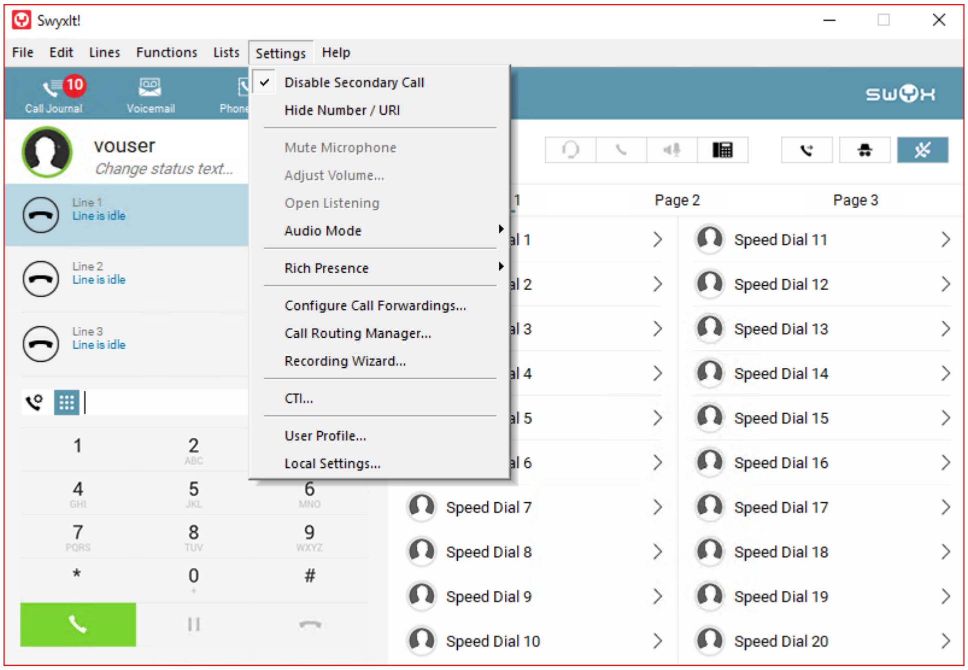
2. Click the Connections Settings tab and add the Server name or IP address and User Name, then click OK.



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

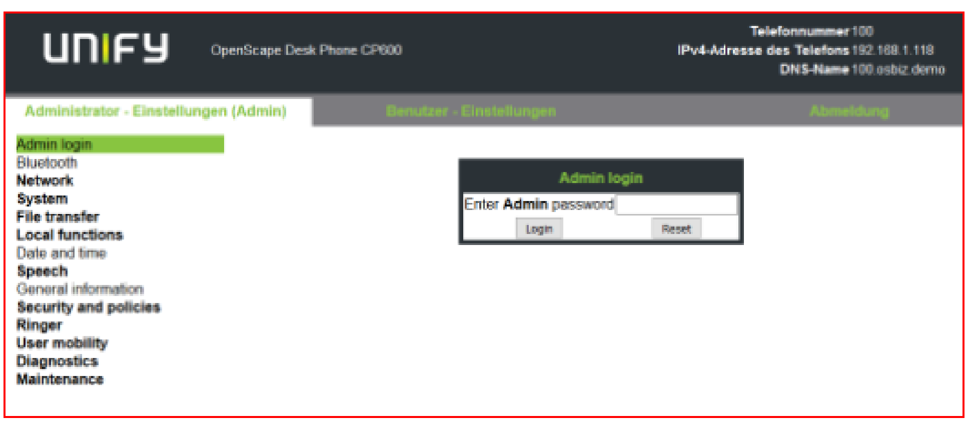
Unify CP 205 comes out of the box as a SIP Phone. Firmware needs to be upgraded before configuring HFA Phone.

Upgrade Using FTP/HTTPS Access Data

The Unify CP 205 comes out of the box as a SIP Phone. You must upgrade the firmware before configuring the 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.

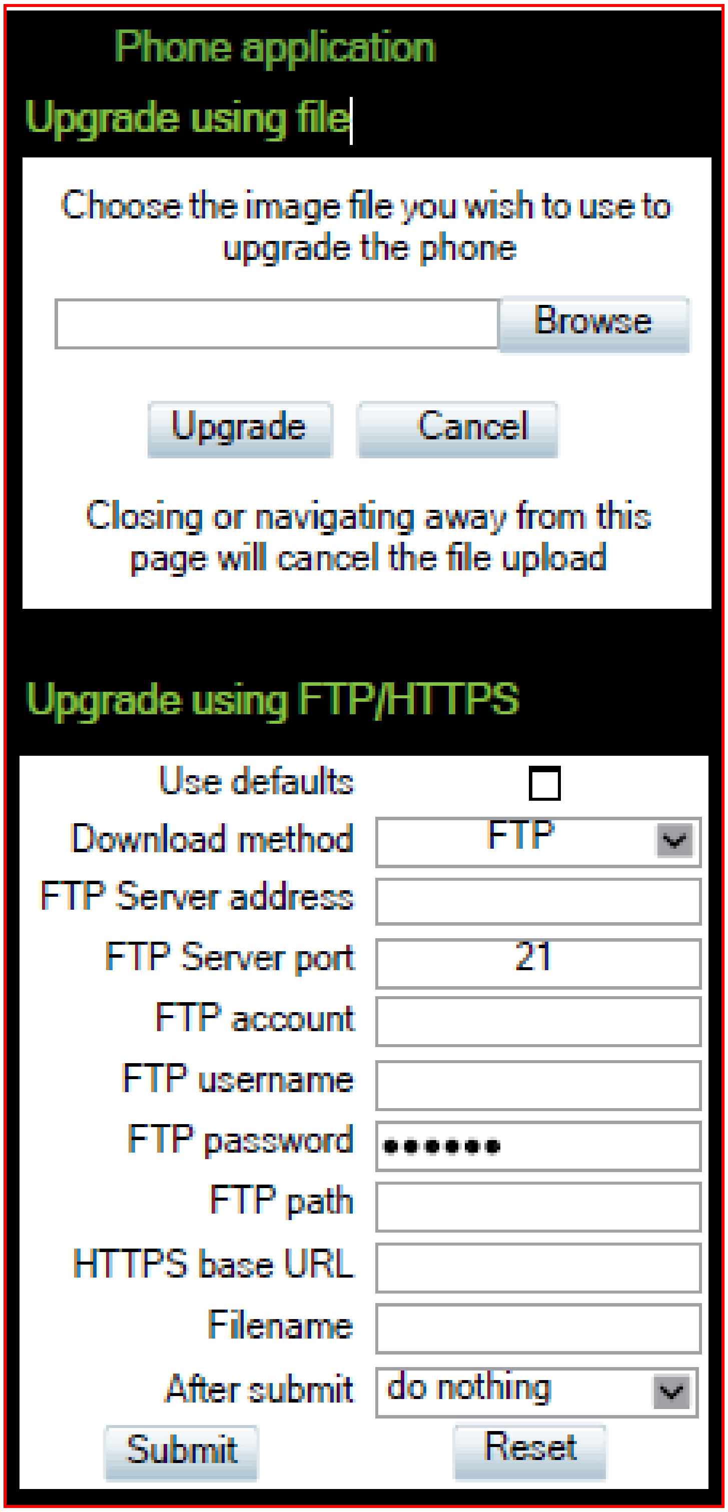
1. Open your web browser and enter the appropriate URL. Example: https://192.168.1.15.



Note: The default password is 123456.

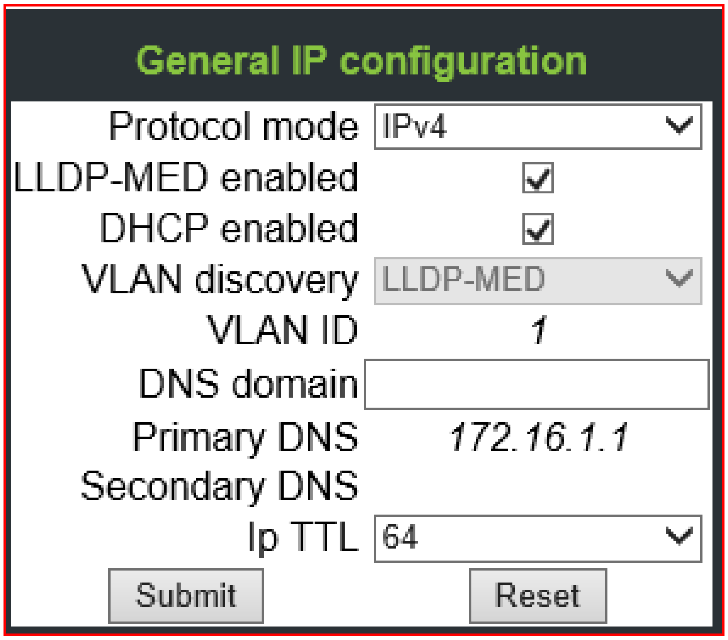
2. From the Administration via Web-Based Management (WBM), select File transfer > Phone application.

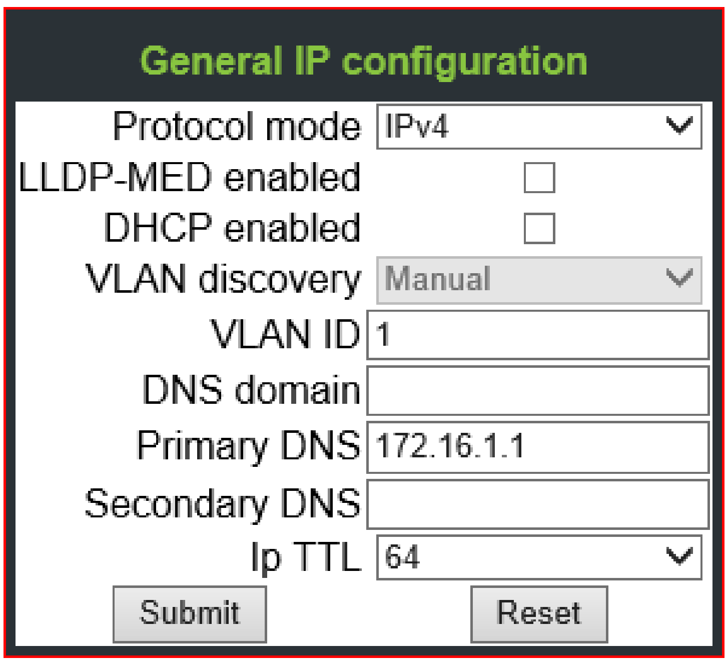
   Click Browse to find the corresponding L62.iso file to upload to the Unify CP205 Phones. Then, click Submit.



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.

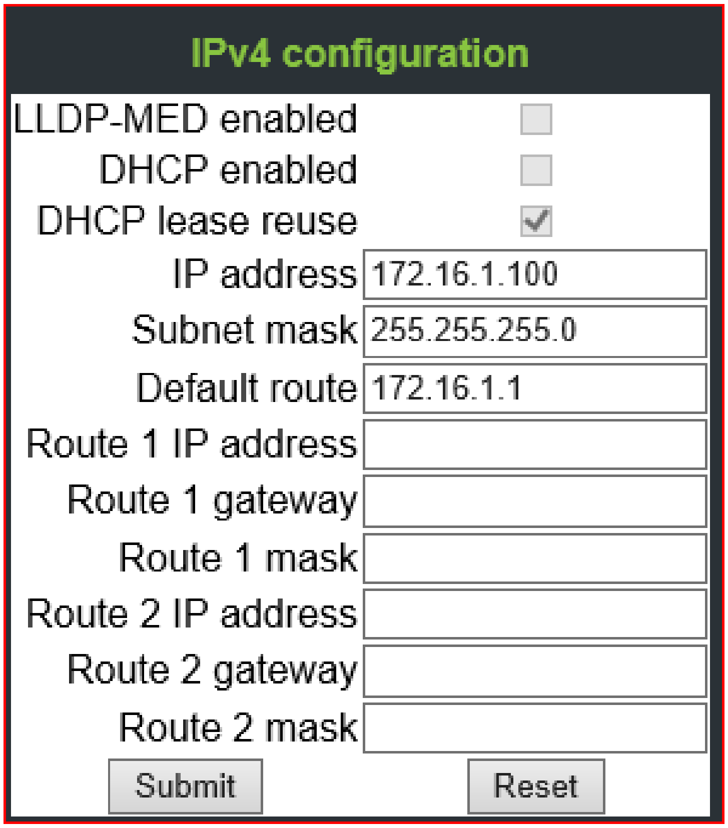
   Choose between DHCP or Static IP assignment.


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
Submit
Reset

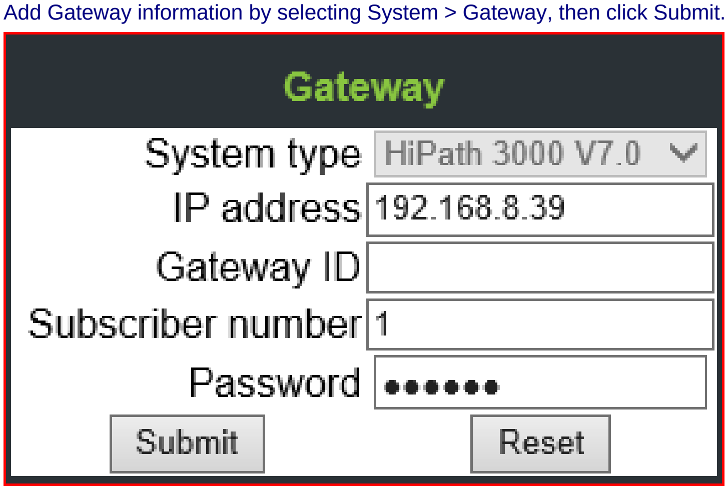

General IP configuration
Protocol mode
IPv4
LLDP-MED enabled
DHCP enabled
VLAN discovery
Manual
VLAN ID
1
DNS domain
Primary DNS
172.16.1.1
Secondary DNS
Ip TTL
64
Submit
Reset

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

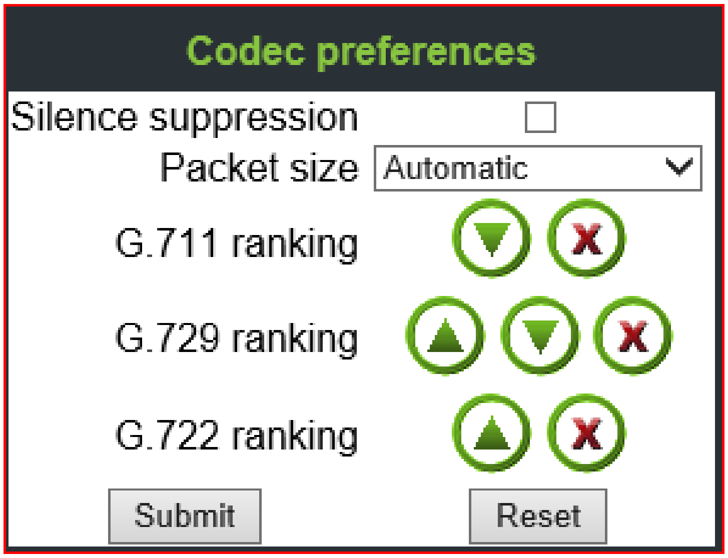
   Note: If DHCP is enabled, the values will be fetched automatically.



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



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



7. Go to the 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 TCP	✓
2	Basic Call Setup & Termination	✓
3	Ringing & Local Ringback Tone	✓
4	Emergency Call 113	✓

5	Cancel Call & Call Rejection	✓
6	Call Forwarding	✓
7	Call Transfer (Attended)	✓
8	Conference Call	✓
9	CLIP	✓
10	CLIR	✓
11	Incoming DTMF	✓
12	Outgoing DTMF	✓

Legend

Supported	✓
Not Supported	✗

Caveats

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

- Fax calls and other tests 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 in order 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.