
EdgeMarc 2900 POE Interop with Vodafone SIP Trunk



Table of Contents

- [Interoperable Carrier](#)
- [Copyright](#)
- [Document Overview](#)
- [Scope](#)
- [Non-Goals](#)
- [Audience](#)
- [Pre-Requisites](#)
- [Product and Device Details](#)
- [Network Topology Diagram](#)
 - [SIP Trunk Deployment Topology](#)
 - [Interoperability Test Lab Topology \(or Call Flow Diagram\)](#)
- [Section-A: EdgeMarc 2900 Configuration](#)
 - [Connectivity](#)
 - [Configuring EdgeMarc ALG](#)
- [Section-B: SwyxWare, SwyxIt, and HFA Phone Configuration](#)
 - [Configuring SwyxPBX](#)
 - [Adding a User](#)
 - [Configuring a SIP Trunk](#)
 - [Configuring the SwyxIt Client](#)
 - [Configuring the HFA Phone](#)
- [Supplementary Services and Features Coverage](#)
- [Caveats](#)
- [Support](#)
- [References](#)
- [Conclusion](#)

Interoperable Carrier



Copyright

© 2021 Ribbon Communications Operating Company, Inc. © 2021 ECI Telecom Ltd. All rights reserved. The compilation (meaning the collection, arrangement and assembly) of all content on this site is protected by U.S. and international copyright laws and treaty provisions and may not be used, copied, reproduced, modified, published, uploaded, posted, transmitted or distributed in any way, without prior written consent of Ribbon Communications Inc.

The trademarks, logos, service marks, trade names, and trade dress (“look and feel”) on this website, including without limitation the RIBBON and RIBBON logo marks, are protected by applicable US and foreign trademark rights and other proprietary rights and are the property of Ribbon Communications Operating Company, Inc. or its affiliates. Any third-party trademarks, logos, service marks, trade names and trade dress may be the property of their respective owners. Any uses of the trademarks, logos, service marks, trade names, and trade dress without the prior written consent of Ribbon Communications Operating Company, Inc., its affiliates, or the third parties that own the proprietary rights, are expressly prohibited.

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

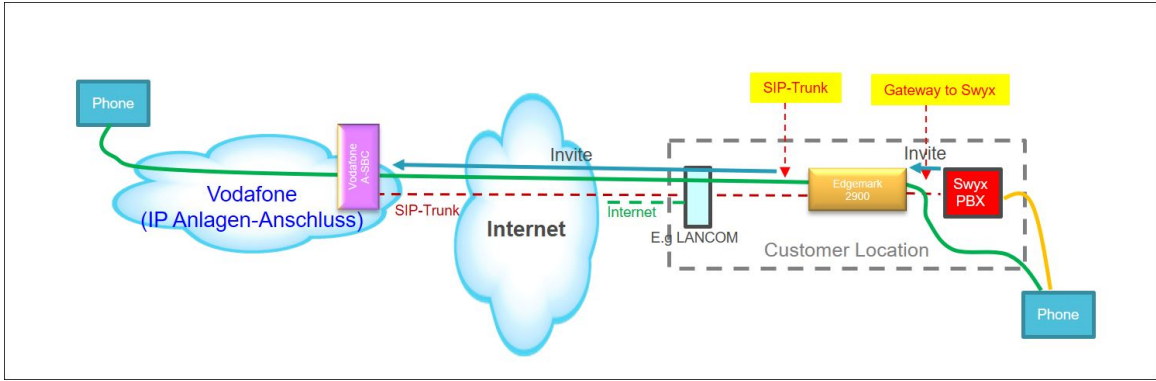
Product and Device Details

	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

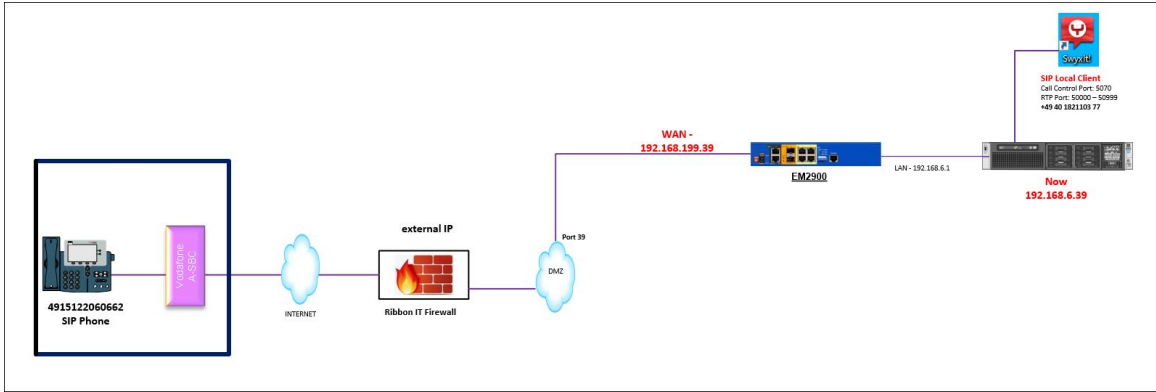
SIP Trunk Deployment Topology

The deployment topology diagram is depicted below.



Interoperability Test Lab Topology (or Call Flow Diagram)

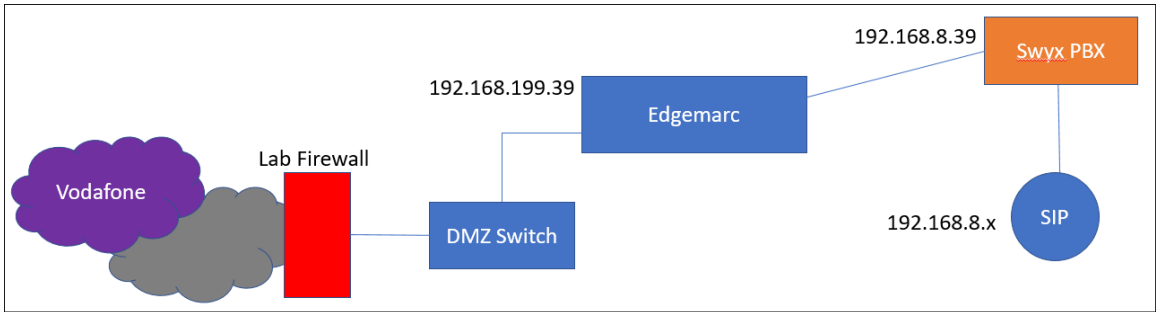
IOT high-level architecture covering call flows and overall topology is depicted below.



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.


VoIP
[Help](#)

Configuration Menu

- + [Admin](#)
- + [Network](#)
- + [Users](#)
- + [Security](#)
- [SD-WAN](#)
- [VoIP](#)
 - [H.323](#)
 - + [SIP](#)
 - [Survivability](#)
 - [Clients List](#)
 - [Test UA](#)
- + [VPN](#)
- [GRE](#)
- + [Switch](#)

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:

Enable LLDP:

LLDP Broadcast Interval (sec):

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:

ALG LAN Interface IPv6 Address:

ALG WAN Interface IP Address:

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:

- 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.

SIP Settings

[Help](#)

SIP protocol settings.

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

SIP Server Transport:

Use Custom Domain:

SIP Server Domain:

List of SIP Servers

Select: [All](#) [None](#) Delete

	Lookup Status	Priority	SIP Server Address/FQDN	Port
<input type="checkbox"/>	●	0	2.207.165.132	5060
<input type="checkbox"/>	●	1	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:

Include UPDATE In Allow:

PRACK Support:

GEOLOCATION Support:

Call Audit Support:

Enable Sub Domain Pass Through support:

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

B2BUA Trunking Configuration Help

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
<input checked="" type="checkbox"/> SwyxPBX	192.168.8.39	65002				UDP

New Entry

Name: Model:

Address(IP/FQDN): Use DNS SRV:

Port: Transport:

Source FQDN:

Username: Password:

Authenticate Registration:

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.

Actions

Name	Send	Prio	Hunt	Header	Refer-To-ReINV
<input checked="" type="checkbox"/> ToPBX	<input checked="" type="checkbox"/>				
<input checked="" type="checkbox"/> ToSIP	<input checked="" type="checkbox"/>			<input checked="" type="checkbox"/>	
<input checked="" type="checkbox"/> Anonymous	<input checked="" type="checkbox"/>			<input checked="" type="checkbox"/>	

New Entry

Name:

Send To: Trunking Device: Client: URI: Response:

Prioritize: Refer to Re-INVITE:

Serial Hunting:

E.164 Conversion rule: Conversion mode:

Header Manipulations:

Header	Value
<input checked="" type="checkbox"/> From	'<sip:' + \$from.uri.user + '@' + \$env.out_intf_host + '>'
<input checked="" type="checkbox"/> Contact	'<sip:' + \$from.uri.user + '@' + \$env.out_intf_host + ':' + \$from.uri.port + '>'
<input checked="" type="checkbox"/> P-Asserted-Identity	'<sip:' + \$from.uri.user + '@' + \$env.out_intf_host + '>'
<input checked="" type="checkbox"/> Privacy	\$privacy.text

Header:

Value:

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.

Match								
Direction	Mode	Def	Called		Calling		Source	Action
			Match	Pattern	Match	Pattern		
<input checked="" type="checkbox"/> Inbound	BothModes		matches	.			Any	ToPBX
<input checked="" type="checkbox"/> Outbound	BothModes				matches	_anonymous	SwyxPBX	Anonymous
<input checked="" type="checkbox"/> Outbound	BothModes		matches	.			SwyxPBX	ToSIP

New Entry

Direction:

Mode:

default

Pattern:

Called Party:

Calling Party:

Source:


Action:

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:

Description:

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

2. Add codes and prefixes, and 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:


Own Area Code:

Prefix for International Calls:

Prefix for Long Distance Calls:

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.

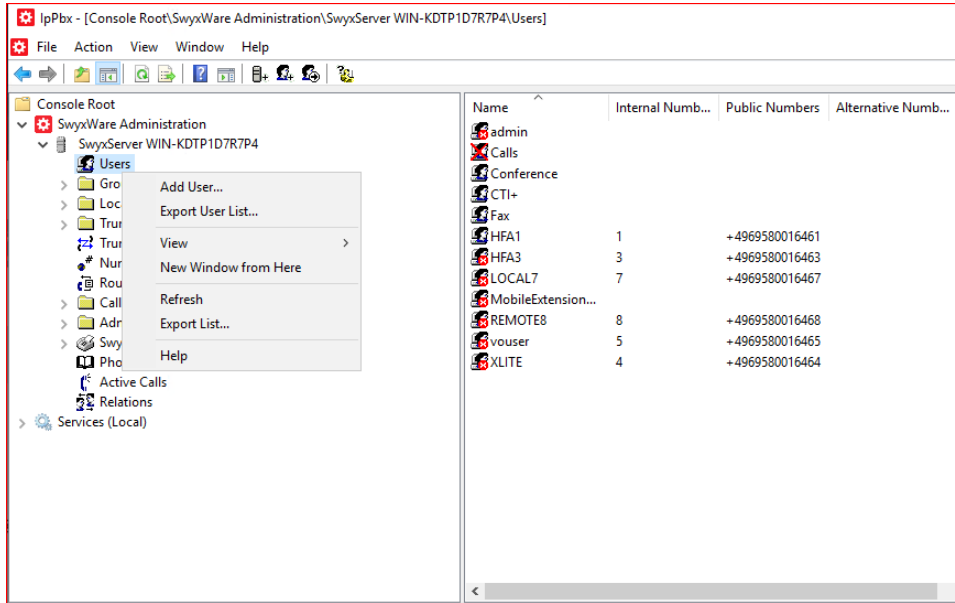
The Internal Number for undeliverable calls defines where incoming calls will be transferred when the called public number is not assigned to a SwyxWare User or Group.

Public Line Access Prefixes:

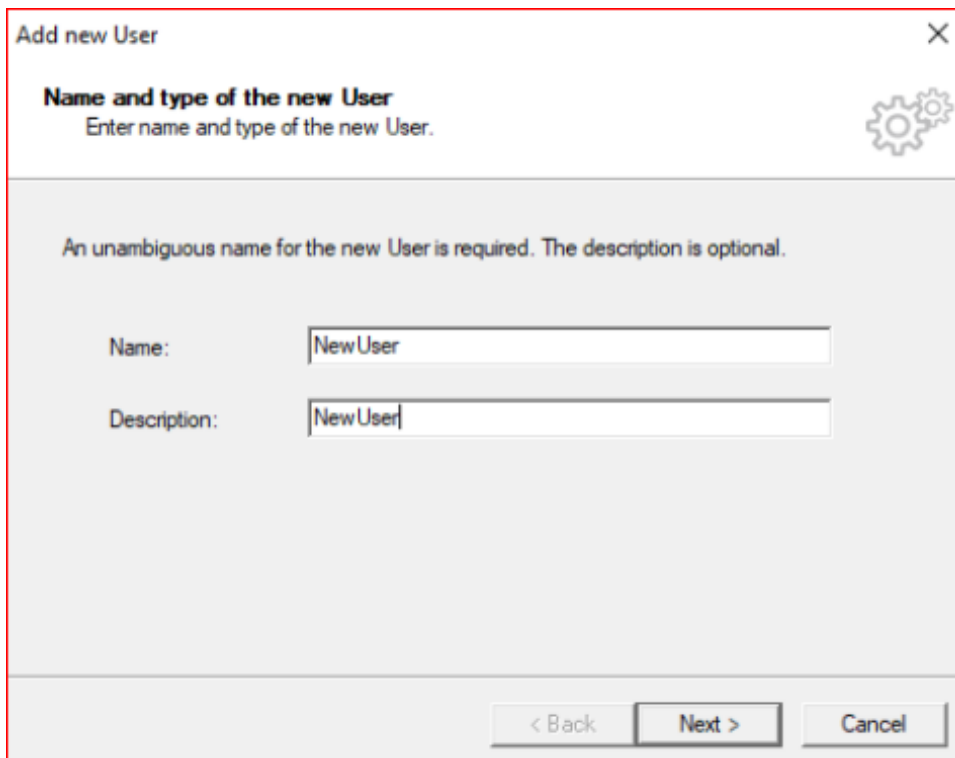
Route undeliverable calls to Internal Number:

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.

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:

Description

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:


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.

Add new User ✕

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:

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

Add new User ✕

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 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: @
Password:
Repeat Password:

User must change password at next logon

8. Create a SIP user and password, 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:


Authentication Mode: ▾

User Name:

Password:
Repeat Password:

9. Create a Swyx Phone Pin, 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:

10. Select a Calling Right, 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:

Description
Default profile allowing calls to all destinations.

11. Select a Feature Profile and click Next.

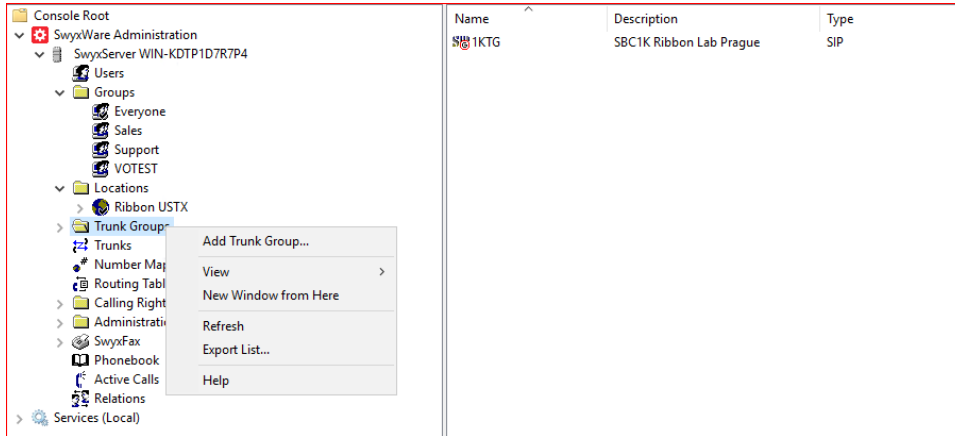
The screenshot shows a dialog box titled "Add new User" with a close button (X) in the top right corner. Below the title is the section "Feature Profiles" with a gear icon and the instruction "Choose Feature Profile." The main content area contains the text: "Feature Profile defines which features a User is allowed to use." and "Please select one of the listed Feature Profiles to define the available features for the User." Below this is a "Feature Profile:" label followed by a dropdown menu showing "Standard". Underneath is a "Description" text box containing the text "Provides the complete feature set offered by the server." At the bottom of the dialog are three buttons: "< Back", "Next >", and "Cancel".

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

The screenshot shows a dialog box titled "Add new User" with a close button (X) in the top right corner. Below the title is the section "Assign properties to the new User" with a gear icon and the instruction "Choose an existing User for transferring his properties to the new User." The main content area contains the text: "You can transfer some properties of an existing User to the new User." and "These properties are mainly SwyxIt! related settings for Skins and ringing sounds. Relationships and group memberships of the existing user are also copied." Below this is the text: "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." There are three radio button options: "Create new User account only", "Create new User account and install sample files for Call Routing" (which is selected and highlighted with a dashed box), and "Create new User account and apply properties of an existing User:". Below the radio buttons is a dropdown menu showing a user icon and the name "admin". At the bottom are two checkboxes: "Open Properties after Finish" and "Send Welcome Mail". At the very bottom of the dialog are three buttons: "< Back", "Finish", and "Cancel".


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.

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:

Description:

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 entering all required parameters in subsequent steps.

Trunk Group Type:

Profile:

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: 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: : 

Realm:

DTMF Mode:

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

< Back Next > Cancel

7. Select the 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:

Encryption Mode:

< Back Next > Cancel

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 the 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:

Description

10. Click Finish.



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

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.

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:

Description:

13. Add the SIP trunk Provider and User Data, and 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 authentication.

SIP Provider:

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
<input type="text" value="49"/>	<input type="text" value="68"/>	<input type="text" value="580016461"/>	<input type="text" value="580016469"/>

15. Add a SIP URI (wild card "*" for any), 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 below 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.711 μ (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)

< Back Next > Cancel

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.

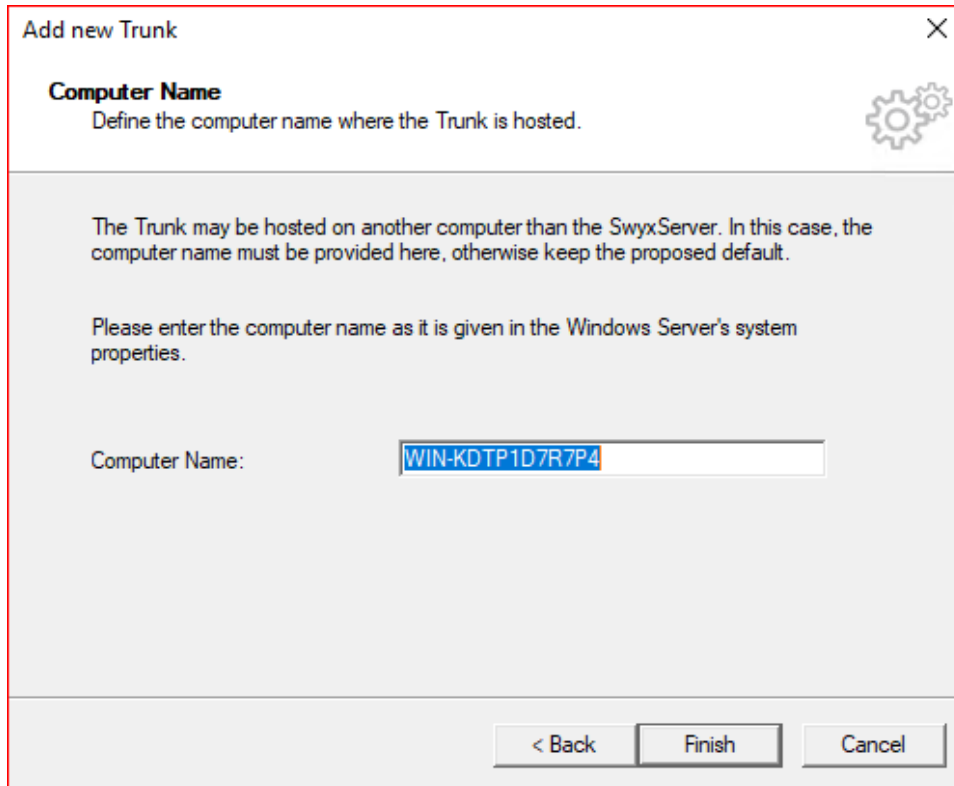
Furthermore the number of simultaneous calls can artificially be limited to reserve (e.g. ISDN) channels or bandwidth for other applications.

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:

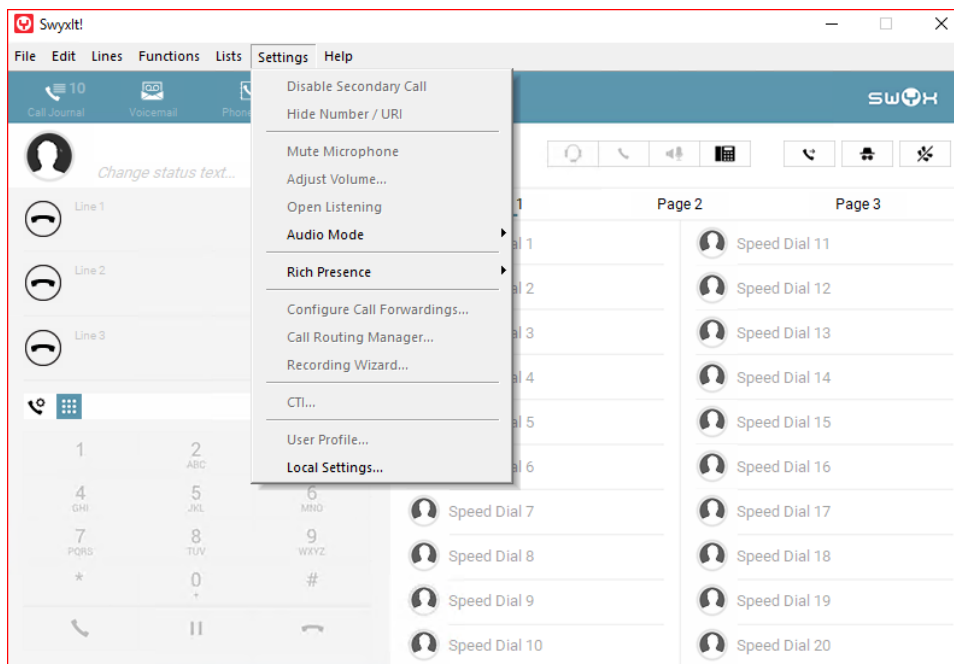
< Back Next > Cancel

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

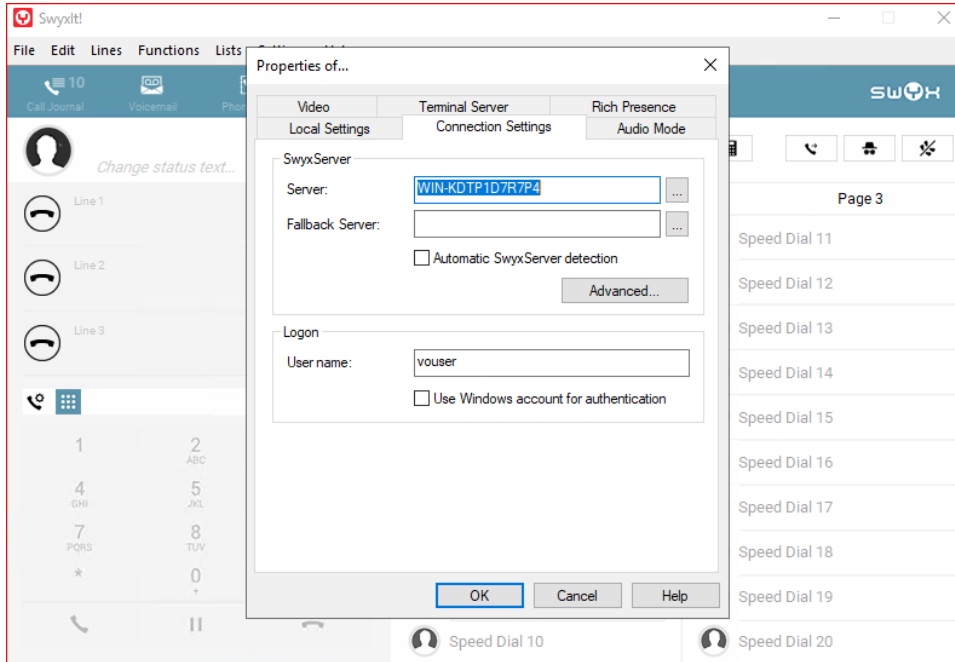


Configuring the SwyxIt Client

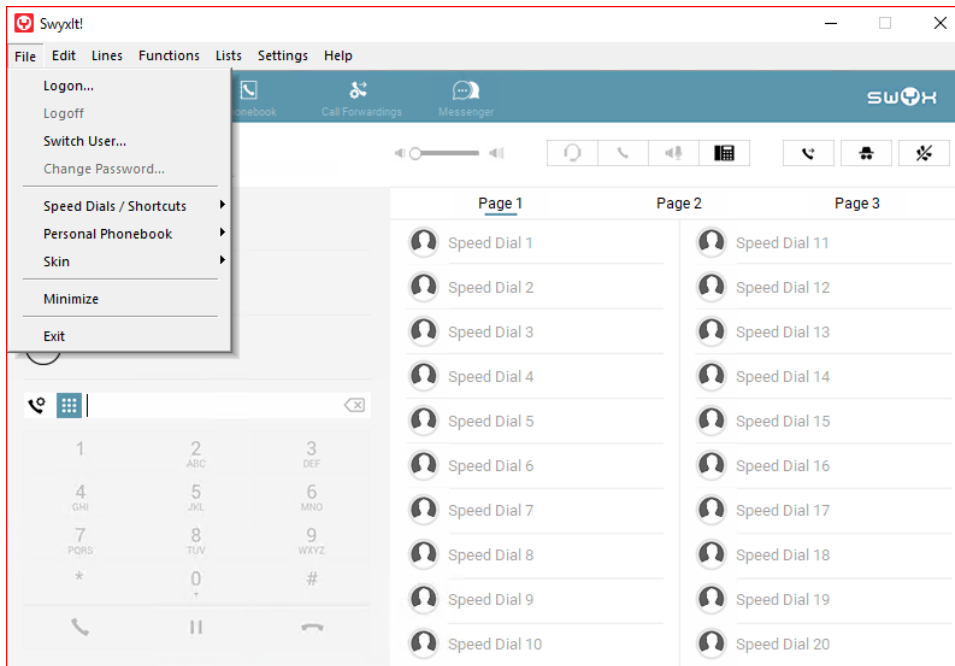
1. Click On Settings and select Local Settings.



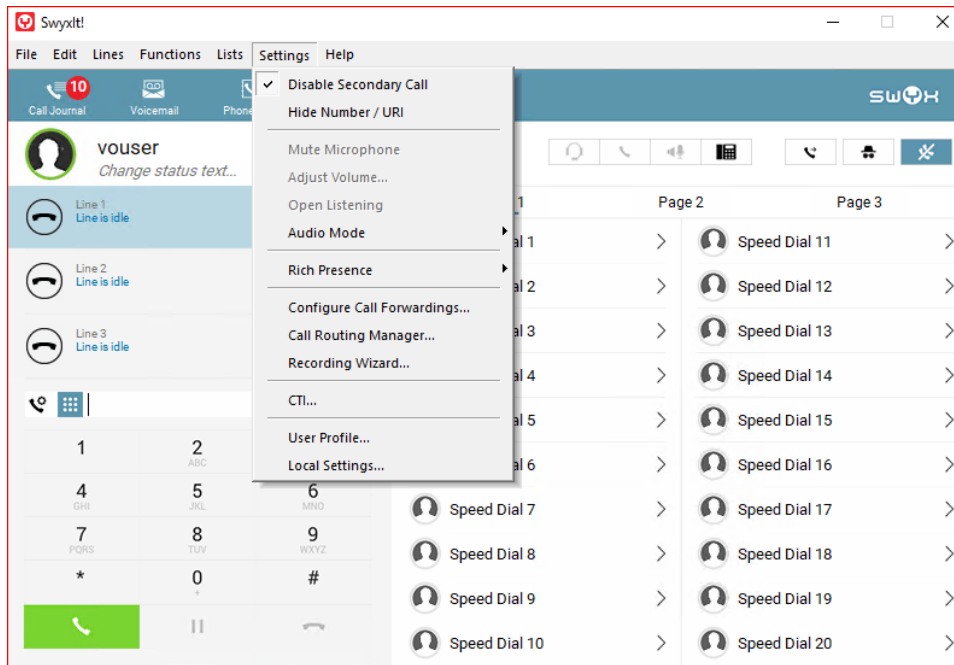
2. Click the Connections Settings tab and add the Server name or IP address and User Name, then click OK.



3. From the File Menu, select Logon.



- 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

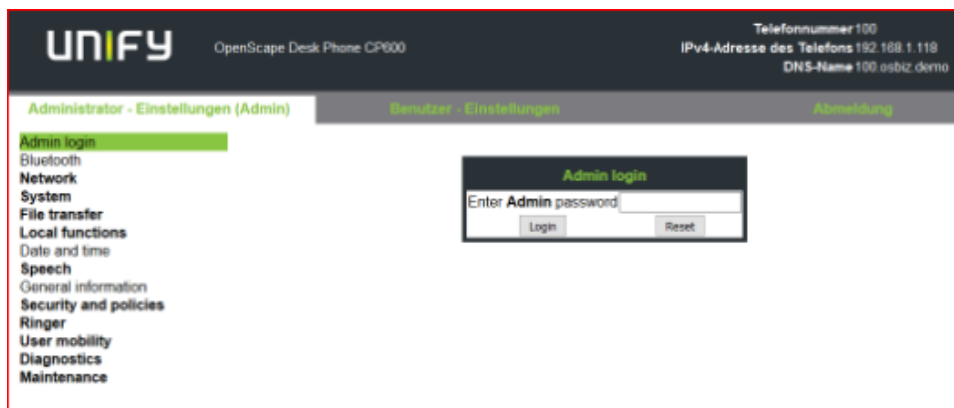
Unify CP 205 comes out of the box as a SIP Phone. Firmware needs to be upgraded before configuring HFA Phone.

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.

- Open your web browser and enter the appropriate URL. Example: <https://192.168.1.15>.



Note: The default password is 123456.

- 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

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

Upgrade using FTP/HTTPS

Use defaults

Download method

FTP Server address

FTP Server port

FTP account

FTP username

FTP password

FTP path

HTTPS base URL

Filename

After submit

- 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	<input checked="" type="checkbox"/>	
DHCP enabled	<input checked="" type="checkbox"/>	
VLAN discovery	LLDP-MED	▼
VLAN ID	1	
DNS domain		
Primary DNS	172.16.1.1	
Secondary DNS		
Ip TTL	64	▼
<input type="button" value="Submit"/>		<input type="button" value="Reset"/>

General IP configuration

Protocol mode	IPv4	▼
LLDP-MED enabled	<input type="checkbox"/>	
DHCP enabled	<input type="checkbox"/>	
VLAN discovery	Manual	▼
VLAN ID	1	
DNS domain		
Primary DNS	172.16.1.1	
Secondary DNS		
Ip TTL	64	▼
<input type="button" value="Submit"/>		<input type="button" value="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.

IPv4 configuration

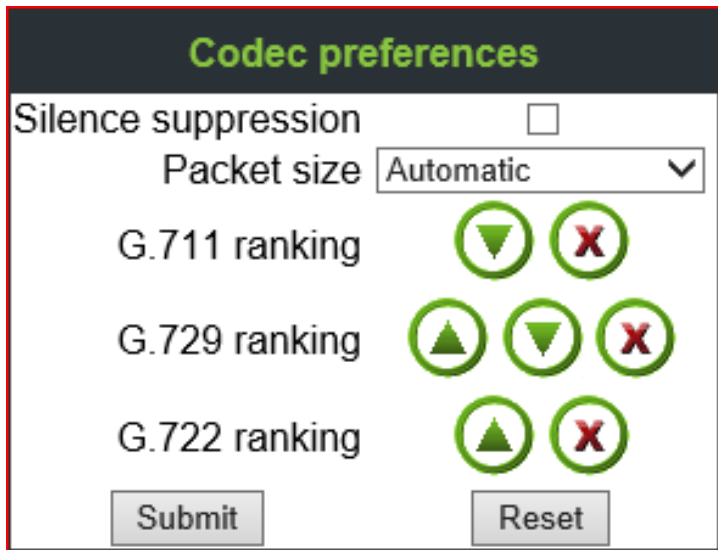
LLDP-MED enabled	<input type="checkbox"/>
DHCP enabled	<input type="checkbox"/>
DHCP lease reuse	<input checked="" type="checkbox"/>
IP address	<input type="text" value="172.16.1.100"/>
Subnet mask	<input type="text" value="255.255.255.0"/>
Default route	<input type="text" value="172.16.1.1"/>
Route 1 IP address	<input type="text"/>
Route 1 gateway	<input type="text"/>
Route 1 mask	<input type="text"/>
Route 2 IP address	<input type="text"/>
Route 2 gateway	<input type="text"/>
Route 2 mask	<input type="text"/>

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

Gateway

System type	<input type="text" value="HiPath 3000 V7.0"/>
IP address	<input type="text" value="192.168.8.39"/>
Gateway ID	<input type="text"/>
Subscriber number	<input type="text" value="1"/>
Password	<input type="password" value="••••••"/>

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	Ringling & 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.

© 2021 Ribbon Communications Operating Company, Inc. © 2021 ECI Telecom Ltd. All rights reserved.

