EdgeMarc 2900 POE Interop with Vodafone SIP Trunk

Cribbon[®]

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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.

noddin 🛟	VoIP	Help
~	VoIP ALG allows the system to recognize and register network d	evices.
Configuration Menu + Admin + Network	Enable ALG Multi-VLAN support: Since VLAN support is enabled, you must select a VLAN for the AL only support one VLAN. ALG LAN using VLAN ID	G to support. The ALG can
+ <u>Users</u> + <u>Security</u> • <u>SD-WAN</u> - <u>VOIP</u>	Enable LLDP: LLDP Broadcast Interval (sec):	✓30
• <u>H.323</u> + <u>SIP</u> • <u>Survivability</u> • <u>Clients List</u> • Test UA	IPv4 only. TFTP Server IP address:	
+ <u>VPN</u> • <u>GRE</u> + <u>Switch</u>	In some cases, the ALG addresses will not correspond to the addr WAN ports. The addresses will be alias addresses that have been general, the user should leave this feature disabled.	resses of the LAN or the configured on the ports. In
	Use ALG Alias IP Addresses:	
	ALG LAN Interface IPv6 Address:	192.100.0.1
	ALG WAN Interface IP Address: ALG WAN Interface IPv6 Address:	192.168.199.39
	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:	✓

- 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.

SIP	Settings			Hel
SIP pr	otocol settings.			
The SI	P Server settings speci	ify the address	and port that all client traffic shall be f	forwarded to.
SIP Se	erver Transport		TCP 🗸	
Use Ci	ustom Domain:			
SIP Se	erver Domain:		ipaahz.ngn.vodafone.de	
		List	of SIP Servers	
Selec	t: <u>All</u> <u>None</u>			Delete
	Lookup Status	Priority	SIP Server Address/FQDN	Port
	۲	0	2.207.165.132	5060
	۲	1	82.113.38.212	5060
٨dd	a new SIP Server	1		I
IP Ac	dress/FODN:			
Port:		- 10		
Add	Keset			
Enable	Multi-homed Outboun	d Proxy Mode:		
Enable	e Transparent Proxy Mo	de:		
Limit (Outbound to listed SIP	Servers:	\checkmark	
Limit I	nbound to listed SIP S	ervers:		
Includ	e UPDATE In Allow:			
PRACK	Support:			
GEOLO	CATION Support:			
Call A	udit Support:		\checkmark	
Enable	Sub Domain Pass Thr	ough support:		

3. The LAN settings are found under VoIP SIP B2BUA. This is a multi-part configuration.

Starting with the PBX SIP trunking configuration:

B2	B2BUA Trunking Configuration								
Thi In the	is page supp order for cha page	oorts only IPv4 nges to this pag	addre e to b	essing. be applie	d, you mu	st click the "Sub	mit" or "Apply Lat	er" button a	nt the bottom of
	Name	Address		Port	Group	Username	Registration	Status	Transport
⊗	SwyxPBX	192.168.8.39	6	5002					UDP .
					N	ew Entry			
	Name:		SwyxF	PBX			Model:	Generic PBX	~
0	Address(IP/	FQDN):	192.16	68.8.39			Use DNS SRV:		
	Port:		65002	2			Transport:	UDP 🗸	
	Source FQDI	N:							
0	Username:						Password:		
	Authenticate	Registration:							
Up	odate								

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.

	Name	Send	Prio	Hunt	Header	Refer-To-ReINV
9	ToPBX	~				
3	ToSIP	✓			✓	
3	Anonymous	~			✓	
				New Entry		
lame		Anonymous				
Send	To:	Trunking	Device:		None 🗸	
		O Client:				
		\bigcirc uri:				
		○ Response	:			
riori	tize:				Refer to Re-INVI	те: 🗆
Serial Hunting:				Add		
					Delete	
E.164	Conversion rule:	None 🗸			Conversion mod	e: Add 🗸
lead	er Manipulations:					
	Header				Value	
🛛 Fr	om	' <sip:' \$from<="" +="" td=""><td>n.uri.user +</td><td>'@' + \$env.ou</td><td>t_intf_host + '>'</td><td></td></sip:'>	n.uri.user +	'@' + \$env.ou	t_intf_host + '>'	
S Co	ontact	' <sip:' \$from<="" +="" td=""><td>n.uri.user +</td><td>'@' + \$env.ou</td><td>t_intf_host + ':' + \$fr</td><td>om.uri.port + '>'</td></sip:'>	n.uri.user +	'@' + \$env.ou	t_intf_host + ':' + \$fr	om.uri.port + '>'
8 P-	Asserted-Identity	' <sip:' \$from<="" +="" td=""><td>n.uri.user +</td><td>'@' + \$env.ou</td><td>t_intf_host + '>'</td><td></td></sip:'>	n.uri.user +	'@' + \$env.ou	t_intf_host + '>'	
🛛 Pr	ivacy	\$privacy.text				
lead	er: Reque	st-URI 🗸				Ad
/alue						

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.

	Direction	Mode	Def	Ca	alled	Calling		Source	Action
				Match	Pattern	Match	Pattern		
8	Inbound	BothModes		matches	10			Any	ToPBX
8	Outbound	BothModes				matches	_anonymous	SwyxPBX	Anonymous
8	Outbound	BothModes		matches	12			SwyxPBX	ToSIP
					New	Entry			
	Direction	: Out	bound	~					
	Mode:	Bot	Modes	~					
0	default								
۲	Pattern:	Cal	ing 🗸						
		Call	ed Party	: matches	~				
		Call	ing Party	r: matches	~		_anony	mous	
	Source:	Swy	rxPBX 🗸]					
	Action	And	nymous	~					

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.	දිදු දු
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.	
< Back Next > C	ancel

2. Add codes and prefixes, and click Next.

Add new Location	×
Location specific codes and prefixes Specify the codes and prefixes which are re	elated to this Location.
The prompted parameters determine how th a SwyxWare User or a Trunk, is interpreted to identify calls that remain in the same area A typical German Location in Berlin would h to '30', International Prefix to '00' and Long	ne destination number of a call, originated by I by the system. This is in particular needed a or county. nave a Country Code set to '49', Area Code Distance Prefix to '0'.
Own Country Code:	1
Own Area Code:	214
Prefix for International Calls:	11
Prefix for Long Distance Calls:	ol
	< Back Next > Cancel

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.	ද ිදුම්
The Public Line Access prefix defines which number has to be dialed to obtain acc to the public network.	ess
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:	
< Back Next >	Cancel

Adding a User

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

 ipPbx - [Console I File Action Vi 	Root\SwyxWare Administration\SwyxSe ew Window Help 🕽 📑 👔 🗊 🖶 🕼 🕵 🐲	rver WIN-KDTP1D7R7P4\Users]			
Console Root Cansole Root SuyxXare Adn SuyxServer Guess Guess Tur Cansole Root Cansole Root	ninistration WIN-KDTP1D7R7P4 Add User Export User List View > New Window from Here Refresh Export List Help Calls ns)	Name Sadmin Calls Conference CTI+ Fax HFA1 HFA3 LOCAL7 MobileExtension REMOTE8 Vouser XLITE	Internal Numb 1 3 7 8 5 4	Public Numbers +4969580016461 +4969580016463 +4969580016467 +4969580016465 +4969580016464	Alternative Numb

2. Add Name and Description, then click Next.

Add new User		×
Name and type of the Entername and type	ne new User le of the new User.	Ś
An unambiguous nam	e for the new User is required. The description is optional.	
Name:	NewUser	
Description:	NewUser	
	< Back Next >	Cancel

3. Select the Location and click Next.

Add new User		×
Location of the new User Please select a Location for the new U	Jser.	بې ې
A Location within SwyxWare defines all lo required public access code, the country	ocation specific settings like the time zone, the and area codes.	
Please select one of the listed Locations	which will be assigned to this User.	
Location:	Ribbon USTX	
- Description		
	< Back Next > Ca	ancel

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 Use Enter the Internal Number, under which the new User will be	er e reachable.	^ر ېږې
To define a Internal Number for this checking if it is already in use. By er will suggest the next free number af	s User, enter the chosen number and click "Verify" for intering a number and clicking "Next unused" the syst fter the given.	em
Uncheck "Show in Phonebook" if y purposes only.	you e.g. want to use the Internal Number for call routir	ng
New Internal Number:	2 Verify Next unused	
	Show in Phonebook	
	< Back Next > Car	ncel

Verify Inte	ernal Number	\times
i	Internal Number '2' is valid and can be assigned to this User.	
	ОК	

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 Us	er's Internal Number will be mapped.
To permit calling this User directly from the pul Internal Number to a Public Number.	blic network, you have to associate the
To do so, choose one of the suggested Public Public Number (canonical format) or SIP URI	c Numbers from the drop-down list, or enter a manually.
Use the "Select" button to obtain an overvi system.	ew of Public Numbers available in the entire
Internal Number: 2	
Associated Public Number:	4969580016462
	< Back Next > Cancel

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

Add new User	×
Terminals Choose which terminals are used.	ζζ ^ζ
A User can make phone calls using different terminals. Check the terminals to be used the new User. The required settings will be configured in the following dialogs.	l by
SwyxIt! and SwyxFax Client	
SIP devices	
SwyxPhone Loox	
Simple User account for call routing. No logins allowed.	
< Back Next > (Cancel

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

Add new User		×
SwyxIt! and SwyxFax Client Lo Define how the new user can lo	o gin Settings ogin with SwyxIt! and SwyxFax Client.	ξĊ ^ĵ
Specify a Windows account, e.g. the same Windows domain or defi login.	when SwyxServer and SwyxIt! or the Swyx ne a User name and password the clients h	Fax Client are in have to use for
Windows User Account:		Browse
✓ User Name and Password	,	
User Login:	NewUser @	
Password:	•••••	
Repeat Password:	*****	
	User must change password at next	logon
	< Back Next >	Cancel

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

Add new User		×
SIP parameters Configure the SIP parameters.		٢ Color Col
To logon via SIP it's necessary to spe In case authentication is enabled you	ecify a unique User ID for each User. I must enter a usemame and a password, too.	
User ID:	NewUser	
Authentication Mode:	<use default="" swyxserver=""></use>	
User Name:	NewUser	
Password:	*****	
Repeat Password:		
	< Back Next >	Cancel

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

Add new User	×
PIN for SwyxPhone Lxxx Enter the PIN.	٢ ^{cc}
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	
Create PIN	
< Back Next > 0	Cancel

10. Select a Calling Right, then click Next.

Add new User		×
Calling Rights Choose Calling Right.		ŚŚ
Calling Rights represent individual call permissi User.	ons or restrictions which can be ass	igned to a
Please select one of the listed Calling Rights to	define the call permissions of the U	Jser.
Calling Right:	call restrictions	•
Description		
Default profile allowing calls to all destin	nations.	
	< Back Next >	Cancel

11. Select a Feature Profile and click Next.

Add new User	×
Feature Profiles Choose Feature Profile.	Ś
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 th	he User.
Feature Profile: Standard	-
Description Provides the complete feature set offered by the server.	
< Back Next >	Cancel

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

Add new User	×	
Assign properties to the new User Choose an existing User for transferring his properties to the new User.	i i i i i i i i i i i i i i i i i i i	
You can transfer some properties of an existing User to the new User.		
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.		
C 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		
< Back Finish	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.

Console Root		Name	Description	Туре
SwyxWare Administration		SH81KTG	SBC1K Ribbon Lab Prague	SIP
SwyxServer WIN-KDTP1D	07R7P4		j	
🔝 Users				
🗸 🧰 Groups				
🕵 Everyone				
🔣 Sales				
🛃 Support				
VOTEST				
Locations				
> 🌄 Ribbon USTX				
> 🔄 Trunk Group	UT 1.0			
Trunks A	dd Trunk Group			
📲 Number Mar Vi	iew >			
e Routing Tabl	ew Window from Here			
> Calling Right				
> Administratio Re	efresh			
> 🥘 SwyxFax Ex	oport List			
Phonebook				
C Active Calls H	elp			
Sections (Lease)				
> 🐝 Services (Local)				



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

Add new Trunk Group		×
Trunk Group Name and De Specify Trunk Group name	scription and description.	$\tilde{\mathbf{C}}$
Enter a unique Trunk Group Group name or Phonebook	o name, i.e. not used otherwise as Trunk name, Us entry.	er name,
Enter the optional descriptio	n that will later on help you identifying this Trunk G	roup.
Trunk Group Name:	SIPTG	
Description:	SIP Trunk Group	
	< Back Next >	Cancel

3. Select the Trunk Group Type and click Next.

Add new Trunk Group		×
Trunk Group Type Specify the type of the Trunk Group and s	select the appropriate profile.	ૢ ૢૢૢૢૺૢ
Select the Type of Trunk Group to be add applicable profile from the second list. If yo for your installation, consult the SwyxWare If you want to add a Trunk Group for a no Profile "Custom". This will allow enetering	ded from the first list and choose the ou are uncertain, which profile is applicable e Administration documentation. n-listed SIP service provider, select the all required parameters in subsequent step	e 08.
Trunk Group Type:	SIP	
Profile:	<customized sip=""></customized>	
	< Back Next > (Cancel

4. Add SIP settings and click Next.

Add new Trunk Group					×
SIP settings Please specify whether SIF	^o registrati	on is enat	oled for this Tr	unk Group.	ζÇ ^ζ
The subsequently prompted provider.	d informati	ion must h	ave been sup	plied by your SI	P service
If your service provider req and enter the registrar's na	uires a SI me or IP a	P registrati address.	on (usual cas	e), enable the cl	heckmark
The SIP account specific i Trunk Group you are curre	nformatior ntly creati	n must be ng.	entered when	you add a Trun	k to the
Enable SIP registration					
Registrar:				:	<u> </u>
Re-registration Interval:	120	÷	seconds		
-					
			< Back	Next >	Cancel

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

Add new Trunk Group				×
SIP Settings Specify SIP settings fo	or this Trunk Group.			ૢૼૢૺ ૽ૺ
The SIP Proxy is the s IP address must have The SIP realm is part composition. The pare	ervice provider's interface been provided. of the SIP addressing mect meter "DTMF Mode" dete	for call contro hanism, i.e. it emines how a	ol. Therefore its is used for SIP a user's keypad	name or URI input is
Dutbound Proxy:	r.		:	Â
Proxy:	192.168.6.1		:	Â
Realm:				
DTMF Mode:	RFC 2833 Event	•		
		< Back	Next >	Cancel

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".	t
STUN Server: Port: 3478	
< Back Next >	Cancel

7. Select the Transport Protocol and click Next.

Add new Trunk Group		×
Encryption Settings Please specify the SIP Trunk Group.	Transport Protocol and the Voice Encryption Mode for this	??? ??
Some SIP providers req the transport protocol w	uire a specific transport protocol. If you choose "Automatic", ill be determined via DNS resolution.	
Voice Encryption can o	only be configured, if "TLS" is selected as transport protocol.	
Transport Protocol:	UDP	
Encryption Mode:	No encryption	
	< Back Next > C	ancel

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.	Ś
Depending on your choice, initial Routing Records will be created. Public Numbers should be added in canonical format (e.g. "+4930123456"), "*" can used as a wildcard.	be
Use Trunks of this Trunk Group	
O for all external calls	
O for all external calls to the following Called Party Number or SIP URI only:	
C For all external calls and all unansigned internal Numbers	
or all external calls and all unassigned internal Numbers	
O for Internal Numbers:	
O Do not create initial Routing Records.	
< Back Next > C	Cancel

9. Select the Location Profile and click Next.

Add new Trunk Group	×
Location Profile Select the applicable Location Profile for this Trunk Group.	र्दुः
A Location within SwyxWare defines all location specific settings like the time zone required public access code, the country and area codes.	, the
Please select one of the listed Locations which will be assigned to this Trunk Group).
Location: Ribbon USTX	•
Description	
< Back Next >	Cancel

Add new Trunk Group		×
	You have successfully completed the Add Trunk Group Wizard.	
O Line	After finishing this Wizard you can assign Trunks to the created Trunk Group.	
11	< Back Finish Cance	

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

Consider Kodd Name Description Type V SwyxWare Administration Name Description Type V SwyxServer WIN-KDTP1D7R7P4 SwyxWare Administration SwyxServer WIN-KDTP1D7R7P4 SwyxServer Win-KDTP1D7R7P	Concola Poot			· · ·	D	-
 WyxWare Administration SwyxServer WIN-KDTP1D7R7P4 Users Groups Everyone Sales Support VOTEST Locations Ribbon USTX Trunk Groups KTG Swy SiP Add Trunk Trunk Groups KTG New Window from Here Calling Calling Calling Admin Delete SwyxFe Refresh Phone Export List 		••		Name	Description	lype
✓ SwyxServer WIN-KDTP1D7R7P4 ✓ Groups ✓ Groups ✓ Sales ✓ Support ✓ Locations ✓ Icutations ✓ Trunk Groups ✓ Trunks ✓ Add Trunk ✓ Numb ✓ Delete ✓ SwyxFer ✓ Export List	Swyxware Adminis	tration				
[©] Users [©] Groups [©] Sales [©] Support [©] VOTEST [©] Iccations [©] Ribbon USTX [©] Trunk Groups [®] Sibon USTX [©] Trunk Groups [®] Numb [®] Numb [®] Calling [®] Admin Delete [®] SwyFi [®] SwyFi [®] SyyFi	✓ SwyxServer WIN	I-KDTP1D7R7P4				
✓ Groups Ø Everyone Ø Sales Ø Support Ø VOTEST ✓ Locations > @ Ribbon USTX ✓ Intruk Groups Ø IKTG Ø Suport	🔝 Users					
Image: Seles Image: Seles Image: Seles Image: VOTEST Image: Seles	🗸 🧰 Groups					
Sales Support Supp	😡 Everyon	e				
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<hr/> • * Numb View > • □ Routin New Window from Here > □ Calling Delete * ● Admin Delete □ Phone € xport List	🔁 Trunks	Add Hulk				
Image: Weight of the second	_ ● * Numb	View	>			
> Calling New Window Hole Here > Admin Delete > SwyxFi Refresh Phone Export List	i Routin	New Window from Hose				
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t Active	D Phone	Ketresh				
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34 Kelatio Properties	S Kelatio	Properties				
> Services (Loca Help	> 🥋 Services (Loca	Heln				
()		(icip		<		

Add new Trunk		×
0	Welcome to the Add Trunk Wizard	
Une t	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.	
1	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. Group name or Phonebook entr	. not used otherwise as Trunk Group name, User r y.	name,
Enter the optional description th	at will later on help you identifying this Trunk.	
Taurk Name:	NewSIPTauck	
Description:	New SIP Trunk	
	< Back Next >	Cancel

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

Add new Trunk		×
SIP Trunk Provider / User Data Specify your account data.		٢
Enter the user identification data a will be used to compose your SIP for authentification.	as provided by your SIP service provider. The user ID address while user name and password will be used)
SIP Provider:	SIP (Customized)	-
User ID:		
User Name:		
Password:		
Repeat Password:		
	< Back Next > Car	ncel

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

Add new Trunk				×
Subscriber Nu Specify Sub	mbers scriber Numbers	5.		1
Enter the su Trunk. If your set of and add the If this Trunk	bscriber number subscriber num other subscribe does not add a	r part of the Public Numbers t ibers is incoherent enter only er numbers later via the Trunk ny Public Numbers to the sys	that are terminated by the first subscriber ni t's properties. term leave all fields e	y this umber empty and
click 'Next'. Note: Count location.	ry Code and Are	a Code have been pre-deter	mined by the Trunk (Group's
Country Code	Area Code	First Subscriber Number	Last Subscriber N	umber
49	68	580016461	- 580016469	
		< Back	Next >	Cancel

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

Add new Trunk	X
SIP URI Specify SIP URI.	20 2
If this Trunk is supposed to handle non-nun service provider) you can enter one of thes Trunk's properties.	neric SIP URIs (e.g. assigned by your SIP e bellow and add other URIs later via the
SIP URIs have the following format:	
sip: <name1> @ <name< th=""><th>2></th></name<></name1>	2>
with <name1> reflecting the user's name an</name1>	d <name2> the realm.</name2>
For convenient input "" can be used as wild address all users in the realm "company.con with the configured realm in the SIP propert	lcard so that *@company.com would n''. The realm field shown below is pre-filled ies but may be overwritten case by case.
URI: sip: *	@ 1
	< Back Next > Cancel

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

Add new Trunk			×
Codecs Select the codecs to be used for data transm	nission.		ૼૢૺ૾ૺ
The selected codec preference and filter def using this Trunk. Therefore the selected cod and the quality of the call. Codecs Preference and Filter	ines the type of ec has an impa	compression for o ct on the used ba	calls ndwidth
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)	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 thisTrunk.	ද ්දුණි
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. <u>c</u> 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 > 0	Cancel

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

Add new Trunk	×
Computer Name Define the computer name where the Trunk is hosted.	20 2
The Trunk may be hosted on another computer than the SwyxServer. In this computer name must be provided here, otherwise keep the proposed default	case, the
Please enter the computer name as it is given in the Windows Server's system properties.	m
Computer Name: WIN-KDTP1D7R7P4	
< Back Finish	Cancel

Configuring the Swyxlt Client

1. Click On Settings and select Local Settings.

😧 Swyxlt!							– 🗆 X
File Edit L	ines Functions	Lists	Settings	Help			
€≣ 10 Call Journal	Voicemail	Phone	Disat Hide	ole Secondary Call Number / URI			ຣພູ⇔ິ່∺
Ω	Change status to	ext	Mute Adju	: Microphone st Volume	Q	\	₹ * %
Line			Oper	Listening	1	Page 2	Page 3
\bigcirc			Audi	o Mode	• al 1	0	Speed Dial 11
			Rich	Presence	al 2	Ω	Speed Dial 12
<u> </u>			Conf	igure Call Forwardings		0	
Line			Call F	Routing Manager	al 3		Speed Dial 13
\bigcirc			Reco	rding Wizard	al 4	Ω	Speed Dial 14
v 📖			CTI			0	Speed Dial 15
1	2		User	Profile	11.5		Speed Diar 15
	ABC		Local	Settings	al 6	Ω	Speed Dial 16
4 GHI	5 JKL		6 MN	n Sp	eed Dial 7	Ω	Speed Dial 17
7 PQRS	8		9		eed Dial 8	0	Speed Dial 18
*	0+		#	n sp	eed Dial 9	Ω	Speed Dial 19
~	11		-	S Sp	eed Dial 10	0	Speed Dial 20

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

😲 Swyxlt!					– 🗆 X
File Edit Lines	Functions Lists	Properties of		>	<
€≣ 10 Call Journal	Voicemail Phon	Video	Terminal Server	Rich Presence	ຣພູ♥ୁ∺
Chai	nge status text	Local Settings SwyxServer	Connection Settings	Audio Mode	1 × # %
Line 1		Server:	WIN-KDTP1D7R7P4		Page 3
•		Fallback Server:	Automatic SuperSequent		Speed Dial 11
				Advanced	Speed Dial 12
Line 3		Logon			Speed Dial 13
0		User name:	vouser		Speed Dial 14
۷			Use Windows account f	or authentication	Speed Dial 15
1	ABC ABC				Speed Dial 16
4 GHI	5 				Speed Dial 17
7 PQRS	8 TUV				Speed Dial 18
*	0		OK Ca	incel Help	Speed Dial 19
~	ļi.		Speed Dial 10	(Speed Dial 20

3. From the File Menu, select Logon.

😧 Swyxlt!					– 🗆 X
File Edit Lines	Functions Lists	Settings Help)		
Logon Logoff		nebook Call F	S 💮 orwardings Messenger		ຣພູ⇔ິ່∺
Switch User Change Passw	rord		4 () 4 (Q 5 4	× # %
Speed Dials / S	Shortcuts 🔸		Page 1	Page 2	Page 3
Personal Phon	ebook		Speed Dial 1	Ω	Speed Dial 11
Minimize	[Speed Dial 2	0	Speed Dial 12
Exit			Speed Dial 3	0	Speed Dial 13
			Speed Dial 4	0	Speed Dial 14
ও 📖		<	Speed Dial 5	0	Speed Dial 15
1.	2 ABC	3 DEF	Speed Dial 6	Ω	Speed Dial 16
4 GHI	5 .JKL	6 MNO	Speed Dial 7	Ω	Speed Dial 17
7 PQRS	8 TUV	9 wxvz	Speed Dial 8	0	Speed Dial 18
*	0+	#	Speed Dial 9	0	Speed Dial 19
<u> </u>	ļ1	-	Speed Dial 10	0	Speed Dial 20

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.

😧 Swyxit!						-	
File Edit Lin	es Functions Lists	Settings Help					
Call Journal	Voicemail Phone	Disable Secon Hide Number	dary Call / URI				ຣພູ⇔ິ່ນ
	ouser hange status text	Mute Microph Adjust Volume	ione	0 5	4	۲	# %
Line 1	dla	Open Listenin	g	1	Page 2	Pa	ge 3
	ule	Audio Mode	•	al 1	> 0	Speed Dial 11	>
Line 2 Line is	dle	Rich Presence		al 2		Speed Dial 12	>
line 3		Configure Call	l Forwardings	al 3		Speed Dial 13	>
C Line is	dle	Recording Wiz	zard			Speed Dial 14	
۳.		СТІ				Speed Dial 14	
	0	User Profile		al 5	> 0	Speed Dial 15	>
1	ABC	Local Settings		al 6	> 0	Speed Dial 16	>
4 GHI	5 JKL	6 MNO	Speed Di	ial 7	> 0	Speed Dial 17	>
7 PQRS	8 TUV	9 WXYZ	Speed Di	ial 8		Speed Dial 18	>
*	0	#	Cread Di	ial 0		Speed Dial 10	, ,
<u>с</u>	Ш	-	Speed Di Speed Di	ial 10		Speed Dial 19	>

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.

UNIFY	OpenScape Desi	k Phone CP800		IPv4-Adre	Telefonnummer 100 sse des Telefons 192, 168, 1, 118 DNS-Name 100, osbiz, demo
Administrator - Einstellun	gen (Admin)				
Admin login Bluetooth Network System File transfer Local functions Date and time Speech General information Security and policies Ringer User mobility Diagnostics Maintenance			Admin login Enter Admin password Login	Reset	

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 navigat page will cance	ing away from this I the file upload						
Upgrade using FTF	/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						

 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 en	abled		\checkmark	
[[DHCP en	abled		\checkmark	
V	LAN disc	overy	LLDP	MED	\sim
	VL/	AN ID		1	
	DNS do	omain[
	Primary	DNS	1	72.16.1.	1
S	econdary	DNS			
	l,	p 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	
lp TTL	64 🗸
Submit	Reset

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			
DHCP enabled			
DHCP lease reuse	\checkmark		
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			
Submit	Reset		

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

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

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	\checkmark
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	\checkmark
12	Outgoing DTMF	\checkmark

Legend

Supported	\checkmark
Not Supported	X

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.

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