

# Ribbon EdgeMarc 2900A R15.7 Interop with SIP Connect: Interoperability Guide

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# Interoperable Vendors

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## Document Overview

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This document depicts the configuration details for Ribbon EdgeMarc 2900A interworking and compliance against Deutsche Telekom SIP Connect solution. This is a general reference document that requires user input during the configuration of Ribbon EdgeMarc 2900A.

This guide contains the following configuration sections:

### [Section A: Ribbon EdgeMarc 2900A Configuration](#)

- Captures general Ribbon EdgeMarc SBC configurations for deploying with Deutsche Telekom SIP Connect solution.

### [Section B: Emulated PBX Configuration](#)

- Captures the Phonerlite configuration which is used as a Emulated PBX.

Deutsche Telekom is a telecommunications company that offers a range of fixed-network services, such as voice and data communication services based on fixed-network and broadband technology. They also sell terminal equipment, other hardware, and services to resellers.

## Non-Goals

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

## Audience

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This is a technical document intended for telecommunications engineers with the purpose of configuring both the Ribbon SBCs and the third-party product.

Steps will require navigating the third-party product as well as the Ribbon product using graphical user interface (GUI) or command line interface (CLI).

Understanding of the basic concepts of TCP/UDP/TLS, IP/Routing, and SIP/RTP/RTCP is needed to complete the configuration and any necessary troubleshooting.



### Note

This configuration guide is offered as a convenience to Ribbon customers. The specifications and information regarding the product in this guide are subject to change without notice. All statements, information, and recommendations in this guide are believed to be accurate but are presented without warranty of any kind, express or implied, and are provided "AS IS". Users must take full responsibility for the application of the specifications and information in this guide.

# Pre-Requisites

The following aspects are required before proceeding with the interop:

- Ribbon EdgeMarc 2900A SBC
- DT Digitalisierungsbox Premium

# Product and Device Details

The following equipment and software were used for the sample configuration provided:

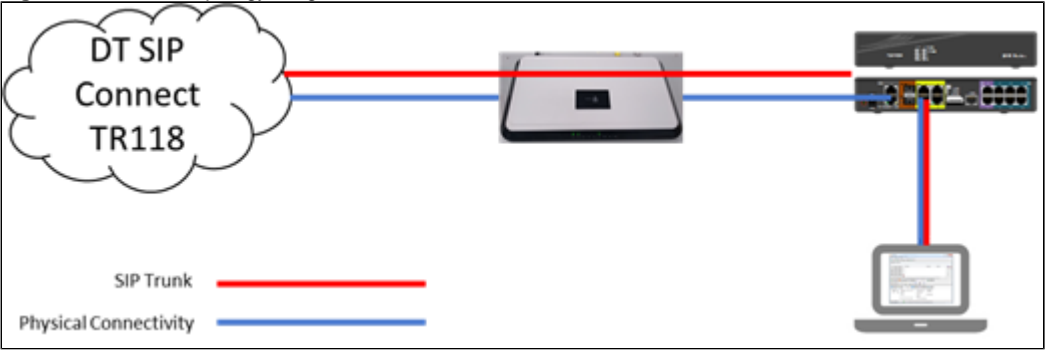
Table 1: Requirements

	Equipment	Software Version
Ribbon Communications	Ribbon EdgeMarc 2900A	V15.7.0
Deutsche Telekom	SIP Connect	TR118
Third-party Equipment	Phonerlite	2.48

# Network Topology Diagram

The following network topology diagram shows connectivity between DT SIP Connect and Ribbon EdgeMarc 2900A.

Figure 1: Network Topology Diagram



# Section A: Ribbon EdgeMarc SBC Configuration


The following EdgeMarc configurations are included in this section:

1. [Login](#)
2. [Network](#)
3. [VoIP](#)
4. [SIP Settings](#)
5. [Survivability](#)
6. [B2BUA Trunking Configuration](#)

## 1. Login

- Login to EdgeMarc and change GUI and SSH Password.

Figure 2: Admin Login


**Admin**
[Help](#)
[Sign Out](#)

### Configuration Menu

- [Admin](#)
  - [Encryption Key](#)
  - [Backup / Restore](#)
  - [Upgrade Firmware](#)
  - [RADIUS Settings](#)
  - [TACACS+ Settings](#)
  - [Services Configuration](#)
  - [System Information](#)
  - [System Analysis](#)
  - [Time Settings](#)
  - [User Commands](#)
  - [File Download](#)
  - [File Server](#)
  - [SD Card](#)
  - [Reboot System](#)
- + [Network](#)
- + [Users](#)
- + [Security](#)
- + [SD-WAN](#)
- + [VoIP](#)
- + [VPN](#)
- + [GRE](#)

**Software Version:**  
Version 15.7.0 -- Mon Dec 16 15:45:54 PST 2019

---

**Hostname:**  
2900A

---

**Model:**  
EdgeMarc 2900A with IPv6 support

---

**Vendor:**  
Ribbon

---

**LAN Interface MAC Address:**  
54:39:68:1B:5B:E5

---

**Registration Status:**  
The ALG feature is registered. View [license key](#).

---

**System:**  
Date : 02/18/2020 15:10:55 UTC  
Erase Button : Enabled

---

**Change Administrative Password:**  
The password of the read-write administrative user can be [changed](#).

---

**Change Read-Only Password:**  
The password of the read-only user can be [changed](#).

- Optional: Connect with EdgeView by navigating to **Admin > Services Configuration**.


**Figure 3:** EdgeView

Current EMPath Management Server:	evd1.emea.rbbn.com
Set EMPath Management Server:	<input type="text" value="evd1.emea.rbbn.com"/>

## 2. Network

- Configure the network to use DHCP on primary WAN port.

**Figure 4:** EdgeMarc Network


**ribbon**

[Help](#)
[Sign Out](#)

## Network

Networking configuration information for the public and private networks.

---

### LAN Interface Settings:

IP Address:

Subnet Mask:

IPv6 Address/Prefix:

Enable VLAN support: ☐

Default VLAN ID:

---

### WAN Interface IPv6 Settings:

Select the type of IPv6 WAN Interface to use:

- ☒ Disabled
- ☐ DHCP
- ☐ Static IP (ethernet)
- ☐ IPv6 in IPv4 Tunnel
- ☐ VLAN

---

### WAN Interface IPv4 Settings:

Select the type of IPv4 WAN Interface to use:

- ☐ Disabled
- ☐ PPPoE
- ☒ DHCP
- ☐ Static IP
- ☐ VLAN

---

To see the IP address given to the WAN port, check the [Network Information page](#).

DHCP client monitor link state ☒

---

### DNS servers:

Note: In case of dynamic links, if the manual override checkbox is not checked the address provided will be used.

Manually set DNS: ☐

Primary DNS Server:

Secondary DNS Server:


### Configuration Menu

- + Admin
- Network
  - + NAT
  - + VLAN
  - + WAN VLAN
  - + 802.1X Supplicant
  - + High Availability
  - + DHCP Relay
  - + DHCP Server
  - + Traffic Shaper
  - + Pass-Through Rules
  - + Subinterfaces
  - + Proxy ARP
  - + Switch Ports
  - + Static Routes
  - + Dynamic DNS
  - + Network Information
  - + Network Restart
  - + Network Test Tools
  - + WAN Failover
  - + Router Advertisement
  - + IP Multicast
- + Users
- + Security
- + SD-WAN
- + VoIP
- + VPN
- + GRE

### 3. VoIP

1. Navigate to **VoIP > B2BUA Options**, and select "Route all SIP signaling through B2BUA".
2. This allows EdgeMarc to act as a back to back user agent and modify signaling exchange according to requirements.

Figure 5: VoIP


**ribbon**

[Help](#)
[Sign Out](#)

**Configuration Menu**

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

**VoIP**

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

---

Enable LLDP:

☒

LLDP Broadcast Interval (sec):

30

---

IPv4 only.

TFTP Server IP address:

---

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

Use ALG Alias IP Addresses:

☐

ALG LAN Interface IP Address:

192.168.1.1

ALG LAN Interface IPv6 Address:

ALG WAN Interface IP Address:

192.168.2.101

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:

☒

Enable Microsoft Feature:

☐

Enable Comfort Noise Generation (CNG):

☐

Enable User-Agent header pass-through:

☐

---

Media Security:

Enable SRTP support:

☐

Enable MKI support:

☐

---

Configure the range of TCP ports to use for handling H.225 and H.245 TCP connections.

H.225/H.245 Port Range:

14085 - 15084

## 4. SIP Settings

1. Navigate to **VoIP > SIP Settings** to configure the SIP settings.
2. Configure registrar FQDN as supplied. There is no need to select custom domain in this instance.
3. Choose required transport protocol and port number for signaling.

Figure 6: SIP



## SIP Settings

[Help](#) [Sign Out](#)

### Configuration Menu

- + Admin
  - + Network
  - + Users
  - + Security
  - + SD-WAN
  - VoIP
    - \* H.323
    - SIP
      - \* ALG
      - \* B2BUA
      - + SIP UA
      - \* SIP GW
      - \* Trunking Group
  - Availability
    - \* Media Server
  - \* Survivability
  - \* Clients List
  - \* Test UA
- + VPN
  - \* GRE

SIP protocol settings.

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

SIP Server Address:

SIP Server Port:

SIP Server Transport:

Use Custom Domain: ☐

SIP Server Domain:

List of SIP Servers:

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: ☐

### Allowed SIP Servers

This is the list of SIP Servers or registrars that are allowed when enabling the "Limit Outbound" (for transparent mode only) and "Limit Inbound" (for transparent as well as non-transparent mode) options. The configured SIP Server(s) above are always included and do not have to be in this list.

#### List of Allowed [Maximum 25] SIP Servers

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

SIP Server Address/FQDN	Port	Transport
-------------------------	------	-----------

The list is currently empty

#### Add a new Allowed SIP Server

IP Address/FQDN:

Port:

Transport:

TCP

Port: 5060
Timeout (minutes): 10

TLS

Port: 5061
TLS Protocol: TLSv1.0
Ciphers String: TLSv1+HIGH:SSLv3:!eN
LAN: Certificate: Default Policy: No check
WAN: Certificate: Default Policy: No check
Exclude sips headers for TLS Transport

NAT Traversal

Warning: This feature is beta and may not function correctly with certain NAT devices
Select the NAT Traversal method to use when the system is behind a NAT device:
☒ Disabled
☐ RFC-3581
☐ STUN

SDP Modifications

SDP Codec Operation: No action
SDP Section that will be modified: audio
Codecs (comma separated list):
Reject when No Match Codec:
Strip Matched Expressions:

SIP Use New Port On Hold Resume:

☒

Priority Numbers

Priority Number 1:
Priority Number 2:
Priority Number 3:
Priority Number 4:

Enable SIP Statistics:

☒

Registration Rate-Pacing parameters are available on the [Survivability page](#).

## 5. Survivability

- Check and confirm SIP connectivity is shown for proxy (and that there is name resolution since this requires DNS SRV lookup) under **VoIP > Survivability**.
- The SIP Server reachability configuration is optional.
- On the same screen, configure registration rate pacing behavior as provided by Softswitch / IP-PBX.

**Figure 7:** Survivability



[Help](#)
[Sign Out](#)

**Configuration Menu**

- + Admin
- + Network
- + Users
- + Security
- + SD-WAN
- VoIP
- + H.323
- + SIP
- + **Survivability**
- + Clients List
- + Test UA
- + VPN
- + GRE

Survivability is a collection of features that enable the system to extend the availability of VoIP services. These features include support for redundant Softswitches/IP PBX's and local call control in the event of WAN link failure, Softswitch/IP PBX failure, or during periods of network congestion that result in loss of connectivity to a remote Softswitch/IP PBX. [Click here for more.](#)

The system is using a dynamic WAN link. Enabling Survivability when using a dynamic WAN link is not a recommended configuration. See the [Help](#) link for more details.

### Current Status

SIP Server Reachability:

	Domain	Name	Address	Port	P	W	Transport	Lost	Rcvd	Status
	reg.sip-trunk.telekom.de	k-ipr-a02.edns.t-ipnet.de	217.0.129.229	5060	10	0	tcp	0	0	Active
	reg.sip-trunk.telekom.de	k-ipr-a01.edns.t-ipnet.de	217.0.129.227	5060	20	0	tcp	0	0	Idle
	reg.sip-trunk.telekom.de	d-ipr-a02.edns.t-ipnet.de	217.0.26.133	5060	30	0	tcp	0	0	Idle

SIP Server Update Received at 13:59:23

Current Call Control is: Remote

### Common Settings

Survivability: Disabled

Time (s) between DNS lookups: 60

### SIP Server Reachability Configuration

The reachability settings control how often messages are sent to the Softswitch/IP PBX and how quickly a Softswitch/IP PBX will be declared unreachable or reachable. The configuration below is used to determine Softswitch/IP PBX reachability for both redundancy and local or remote call control functions.

**Regular Proxy Reachability Detection**

**SIP Keepalive Messages:**

Enable keepalive messages for active server ☐

Time between Keepalive messages (sec.): 5

Number of missed messages to declare alarm: 5

Number of received messages to clear alarm: 10

Interpret error code as success: 403

No-response backoff algorithm: Regular

Maximum backoff interval (sec.): 40

Reachability holdoff (sec.): 0

Ignore holdoff when local ☐

**SIP Requests:**

#### Note

If you do not have SIP Server Redundancy Configuration (below) enabled and DNS SRV lookup, you will not see the multiple entries shown in the screenshot above.

**Figure 8:** SIP Server Redundancy Configuration

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### SIP Server Redundancy Configuration

Redundancy allows the DNS server to give multiple SIP Server names in the answers to SRV lookups. Each server will be monitored using periodic messages and the highest priority answer which is currently reachable will be used for signaling.

Enable SIP server redundancy: ☒

Enable forward next REGISTER ☐

Enable sticky failover mode ☐

Enable SRV Lookup ☒

Enable 503 response for SUBSCRIBE with transparent mode after server failover ☐

### Local Call Control Configuration

Number of digits for local dialing:

Enable Shared Call: ☐

Request Subscriber Information from SIP Server: ☒

Maximum Registration Expiry (s):

Local Mode Indicator:

Local Mode Message:

Pass Through Mode: ☐

Codec Choice:

### Sip Registration Control

**Expires Override**  
The Expires Override settings allow you to configure whether to override the expires values from the phone or the soft-switch in order to modify the registration expiration time.

Enable Phone Expires Override: ☐

Phone Expires Override (s):

Enable Soft-Switch Expires Override: ☐

Softswitch/IP PBX Expires Override (s):

**Registration Rate-Pacing**  
The Registration Rate-Pacing settings allow you to configure the rate that REGISTER messages will be forwarded to the Softswitch/IP PBX.

Rate-Pacing behavior:

Rate-Pacing interval (s):

**Send Deregister after Server Failover**

Enable Sending Deregister after Server Failover: ☐

De-Register Response Expires value (s):

## 6. B2BUA Trunking Configuration

1. Navigate to **VoIP > B2BUA** to configure SIP Trunk on LAN to PBX and WAN to DT Registrar.
2. In the test setup, PBX on LAN was the Phonerlite client on the laptop that had its IP address allocated by EM acting as DHCP server.

**Figure 9:** B2BUA Trunk Configuration

**ribbon** **B2BUA Trunking Configuration** Help Sign Out

**Configuration Menu**

- + Admin
- + Network
- + Users
- + Security
- + SD-WAN
- VoIP
  - + H.323
  - SIP
    - + ALG
    - + B2BUA
    - + SIP UA
    - + SIP GW
    - + Trunking Group Availability
    - + Media Server
    - + Survivability
    - + Clients List
    - + Test UA
- + VPN
- + GBE

**Trunking Devices**

Name	Address	Port	Group	Username	Registration Status	Transport
LANPEX	192.168.1.150	5060				UDP

[New Entry](#)

Name:  Model: Generic PBX

Address(IP/FQDN):  Use DNS SRV: ☐

Port: 5060 Transport: UDP

Source FQDN:

Username:  Password:

Authenticate Registration: ☐

[Update](#)

**Credentials and Registration**

AOR	Auth-User	Password	Registrar	Status	Transport
sip:+4922842223470@sip-trunk.telekom.de	551129822478	is set	reg.sip-trunk.telekom.de:5060	OK	TCP
default	551129822478@sip-trunk.telekom.de	is set			

[New Entry](#)

**Credentials**

Username:  Auth-User:

Edit Password: ☐

Password:

Confirm Password:

Use as default: ☐

**Registrar**

☒ Don't Register

☐ Default SIP Proxy

- The first set of credentials is for the REGISTER challenge to register the SIP trunk.

**Figure 10: Credentials and Registration-1**

**Credentials and Registration**

AOR	Auth-User	Password	Registrar	Status	Transport
sip:+4922842223470@sip-trunk.telekom.de	551129822478	is set	reg.sip-trunk.telekom.de:5060	OK	TCP
default	551129822478@sip-trunk.telekom.de	is set			

[New Entry](#)

**Credentials**

Username: +4922842223470 Auth-User: 551129822478

Edit Password: ☐

Password:

Confirm Password:

Use as default: ☐

**Registrar**

☐ Don't Register

☐ Default SIP Proxy

Custom URI Domain:

☒ Domain: sip:sip-trunk.telekom.de

Address (optional): reg.sip-trunk.telekom.de Port: 5060

Transport: TCP

**Register Options (Optional)**

Default Expires:  sec. Renew interval:  %

[Update](#)

- The next part is for the INVITE challenge when trying to place calls over the SIP trunk.

**Figure 11: Credentials and Registration-2**

### Credentials and Registration

AOR	Auth-User	Password	Registrar	Status	Transport
✖ sip:+4922842223470@sip-trunk.telekom.de	551129822478	is set	reg.sip-trunk.telekom.de:5060	OK	TCP
✖ default	551129822478@sip-trunk.telekom.de	is set			

*New Entry*

---

#### Credentials

Username:  Auth-User: 551129822478@sip-trunk

Edit Password: ☐

Password:

Confirm Password:

Use as default: ☒

---

#### Registrar

☒ Don't Register

☐ Default SIP Proxy

Custom URI Domain:

Domain:

Address (optional):  Port:

Transport:

---

#### Register Options (Optional)

Default Expires:  sec. Renew interval:  %

- Action to send incoming calls to LAN PBX.

Figure 12: Actions 1

### Actions

Name	Send	Prio	Hunt	Header	Refer-To-ReINV
✖ SENDTOLANPBX	✓				
✖ ADDPPHEADER	✓			✓	

*New Entry*

---

Name:

Send To: ☒ Trunking Device: LANPBX

☐ Client:

☐ URI:

☐ Response:

Prioritize: ☐

Serial Hunting:

Refer to Re-INVITE: ☐

E.164 Conversion rule:  Conversion mode:

---

#### Header Manipulations:

Header	Value
Header: Request-URI	<input type="text"/>

- Action for adding P-Preferred Identity Header to all calls going to the DT SIP Trunk (only needed if PBX is not already adding this).

Figure 13: Actions 2

### Actions

Name	Send	Prio	Hunt	Header	Refer-To-ReINV
SENDTOLANPBX	✓				
ADDPPIHEADER	✓			✓	

New Entry

Name: **ADDPPIHEADER**

Send To: **\* Trunking Device:** **None**

☐ Client:   
☐ URI:   
☐ Response:

Prioritize: ☐ Refer to Re-INVITE: ☐

Serial Hunting:

E.164 Conversion rule: **None** Conversion mode: **Add**

#### Header Manipulations:

Header	Value
<b>P-Preferred-Identity</b>	<b>'sip:' + \$from-uri.user + '@sip-trunk.telekom.de;user=phone'</b>

Header: **Request-URI**

Value:

- Match condition for sending all incoming calls to the LAN PBX.

Figure 14: Match 1

### Match

Direction	Mode	Def	Called		Calling		Source	Action
			Match	Pattern	Match	Pattern		
Inbound	BothModes	✓					Any	SENDTOLANPBX
Outbound	BothModes		matches	.			Any	ADDPPIHEADER

New Entry

Direction: **Inbound**

Mode: **BothModes**

**\* default**

☐ Pattern: **Called**

Called Party: **matches**

Calling Party: **matches**

Source: **Any**

Action: **SENDTOLANPBX**

- Action to add P-Preferred Identity Header to outgoing calls as required from LAN PBX to DT Trunk.

Figure 15: Match 2

Match									
Direction	Mode	Def	Called		Calling		Source	Action	
			Match	Pattern	Match	Pattern			
Inbound	BothModes	✓					Any	SENDTOLANPBX	
Outbound	BothModes		matches				Any	ADDPHEADER	
New Entry									
Direction:		Outbound							
Mode:		BothModes							
default									
Pattern:		Called							
Called Party:		matches							
Calling Party:		matches							
Source:		Any							
Action:		ADDPHEADER							
Update									

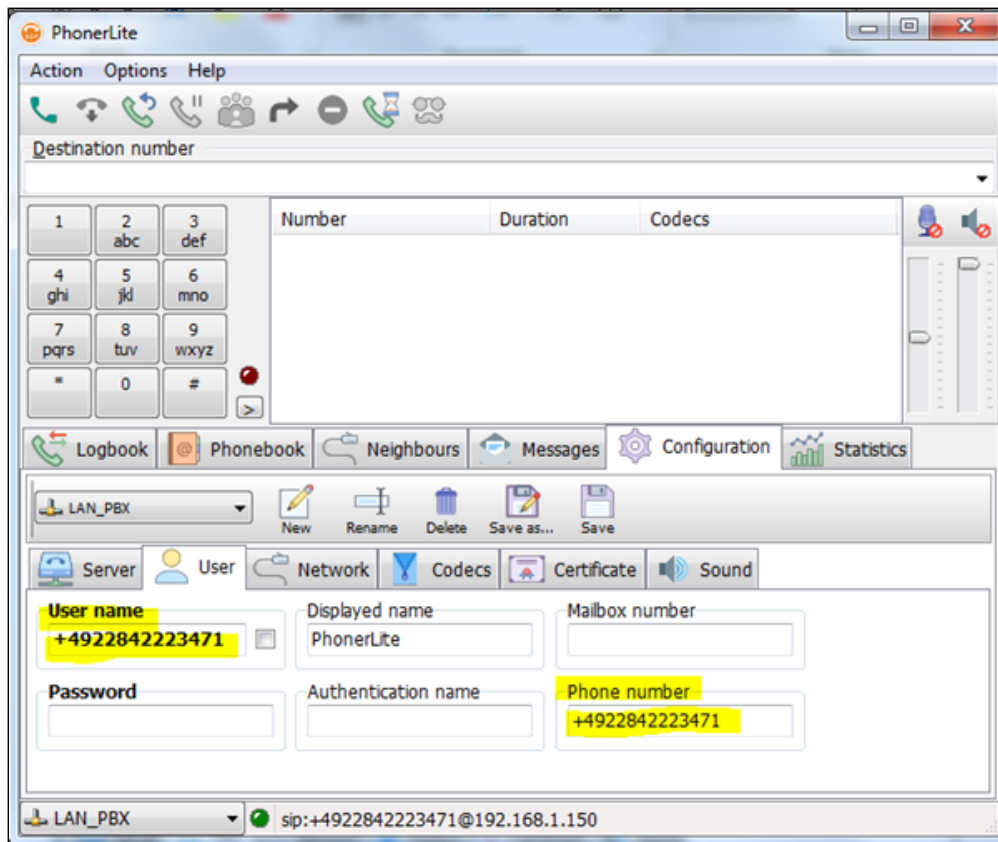
## Section B: Emulated PBX Configuration

Phonerlite is used as an Emulated PBX in the current setup.

Figure 16: Phonerlite 1

The screenshot shows the PhonerLite application window with the 'Configuration' tab selected. The interface includes a menu bar (Action, Options, Help), a toolbar with various call-related icons, and a 'Destination number' input field. Below these are several tabs: Logbook, Phonebook, Neighbours, Configuration (active), and Statistics. The Configuration tab contains a 'Server' section with a dropdown menu set to 'EM\_LANPBX'. Below this are fields for 'Proxy/Registrar' (192.168.1.1), 'Register' (checked), and 'MWI' (checked). There is also a 'STUN server' field and a 'Domain/Realm' field. At the bottom, a status bar shows the current SIP address: 'sip:+4922842276160@192.168.2.118;transport=tcp'.

Figure 17: Phonerlite 2



## Supplementary Services and Features Coverage

The following checklist depicts the set of services/features covered through the configuration defined in this Interop Guide.

Sr.No.	Supplementary Features/Services	Coverage
1	Basic Registration	✓
2	Basic Inbound Call	✓
3	Basic Outbound Call	✓

### Legend

Item	Definition
✓	Supported
✗	Not Supported
N/A	Not Applicable

## Support

For any support related queries about this guide, contact your local Ribbon representative, or refer to the following details:

- 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 and solutions, visit: <https://ribboncommunications.com/products>

For detailed information about Deutsche Telekom products and solutions, visit: <https://www.telekom.com/>

## Conclusion

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This Interoperability Guide describes a successful configuration covering Deutsche Telekom SIP Connect interop involving the Ribbon EdgeMarc SBC. All the necessary features and serviceability aspects stand covered as per the details provided in this interoperability document.

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