



# Innovaphone Virtual Appliance

SIP Trunk configuration guide for Swisscom  
Enterprise SIP

## Enterprise SIP – SIP Trunk configuration guide

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## 1 Introduction

This document provides a summary of required hardware, software, list of supported features, limitations and describes the configuration necessary on Innovaphone VA version 12r1 dvl IPVA [12.0594] to interoperate with Swisscom Enterprise SIP.

## 2 SIP Trunk Network Architecture

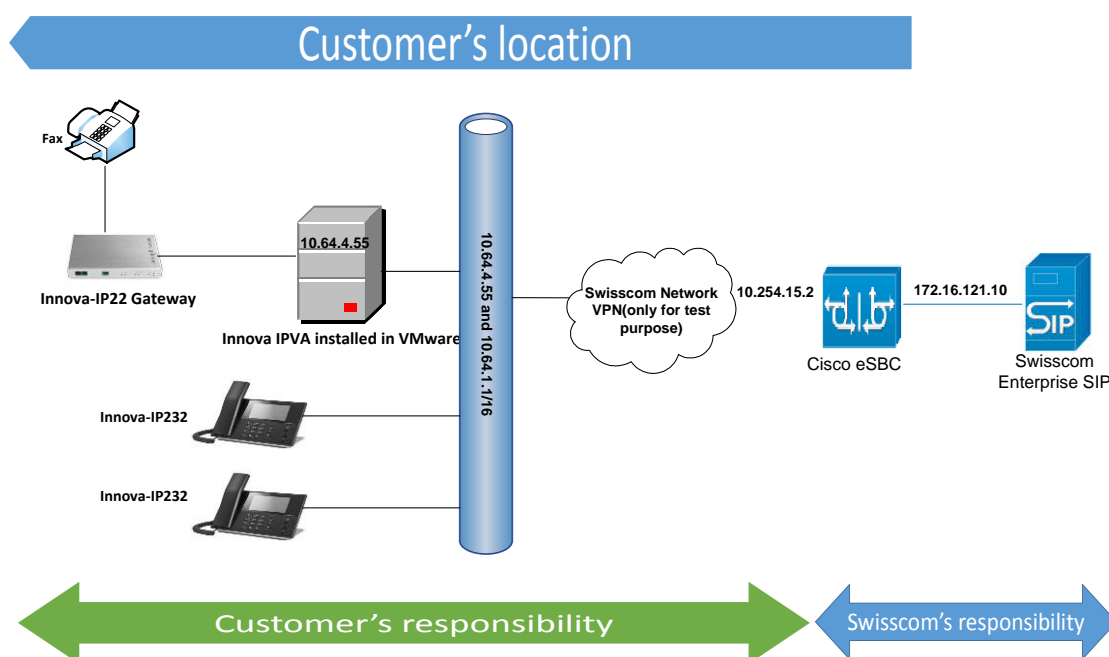


Figure 1

### 2.1 Hardware Components

The following hardware components that were used for homologation purpose:

- Cisco UCS-C240-M3S VMWare Host
- Innovaphone VA, IP232 phones
- Innovaphone IP22 Gateway

### 2.2 Software Requirements

The following software was used for homologation:

- Cisco UCS-C240-M3S VMware vSphere Image Profile: ESXi-5.5.0-1331820-standard
- Innovaphone VA version 12r1 dvl IPVA [12.0594], Boot code [1000]
- Innovaphone IP22 Gateway Version 11r2 dvl IP22[11.3236], Boot code [113236]

## **3 SIP Trunk Features**

### **3.1 Features Supported**

Below are the features that were tested:

- Basic inbound and outbound calls
- Long duration calls
- DTMF after call connect
- DTMF before call connect
- Calls to busy subscriber
- Calls to early media phone
- International calls
- Calls to short numbers
- Calls with special arrangement (spoofing calling party number with +41800XXXXXX/+4179XXXXXX)
- CLIP—Calling Line Identification Presentation
- CLIR—Calling Line Identification Restriction
- Attended call transfer
- Blind call transfer
- CFU—Call Forwarding Unconditional
- CFB—Call Forwarding Busy
- CFNA—Call Forwarding No Answer
- Call hold and resume
- Conference call
- Inbound/outbound FAX with T.38
- Inbound/outbound FAX with G.711 Pass-through

### **3.2 Features Not Supported**

- Geo Location support on emergency calls
- Modem Voice Band Data Mode (not tested)

## 4 Caveats and known Restrictions

These are the known limitations, caveats, or integration issues:

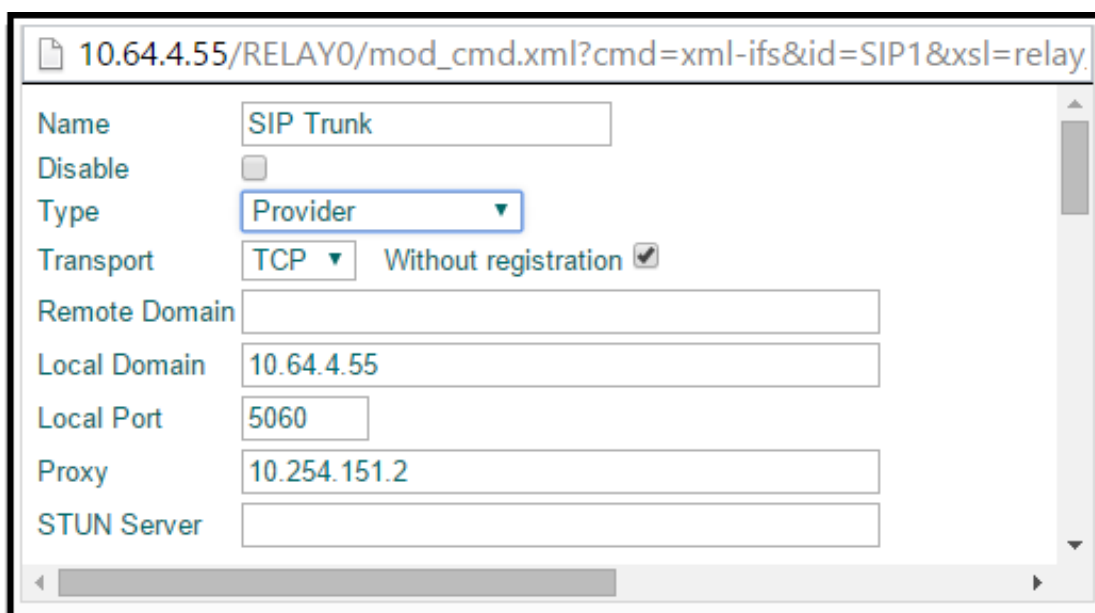
- PBX sends PPI header in the sip provisional response (180 ringing) (this issue will be fixed in a next version, so that PAI (instead of PPI) will be also used for 180 ringing).

## 5 IPVA Configuration

### 5.1 SIP Trunk Configuration

Below is the procedure to setup SIP Trunk on the PBX.

1. Login into the IPVA.
2. Navigate to **Gateway → SIP → Interface → SIP1**
3. Select **Type** as **Provider** in the drop down and **Transport Type** as **TCP**.



Name	SIP Trunk
Disable	<input type="checkbox"/>
Type	Provider
Transport	TCP Without registration <input checked="" type="checkbox"/>
Remote Domain	
Local Domain	10.64.4.55
Local Port	5060
Proxy	10.254.151.2
STUN Server	

4. Local Domain - IP address/Domain name of the PBX, In this case: **10.64.4.55**
5. Local Port - PBX SIP listening Port, In this case: **5060**
6. Proxy IP - IP address of the Remote of the Trunk, In this case: **10.254.15.2**
7. Trunk does not require registration and therefore the option without registration is checked.

**Media Properties**

General Coder Preference **G711A** Framesize [ms] **20** Silence Compression ☐ Exclusive ☒

Local Network Coder **G711A** Framesize [ms] **20** Silence Compression ☐

Enable T.38 ☒ No DTMF Detection ☐ Media-Relay **On** Video ☐ No ICE ☒

SRTCP Cipher **AES128/32** SRTCP Key Exchange **No encryption**

Record to (URL)

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**SIP Interop Tweaks**

Proposed Registration Interval [s]

Accept INVITE's from Anywhere ☐

Enforce Sending Complete ☐ (affects outgoing SIP calls only)

No Video ☐

To Header when Sending INVITE **Called Party** (affects outgoing SIP calls only)

From Header when Sending INVITE **CGPN in user part of URI**

Identity Header when Sending INVITE **CGPN in user part of URI**

Reliability of Provisional Responses **Supported** (affects outgoing SIP calls only)

Microsoft Presence Format ☐

## Media Properties

8. Under the Media Properties, select the General Coder Preference as **G711A**.
9. **Enable T.38** for Fax support.
10. Media-Relay: Select Media-relay **ON**.

## Number Mapping

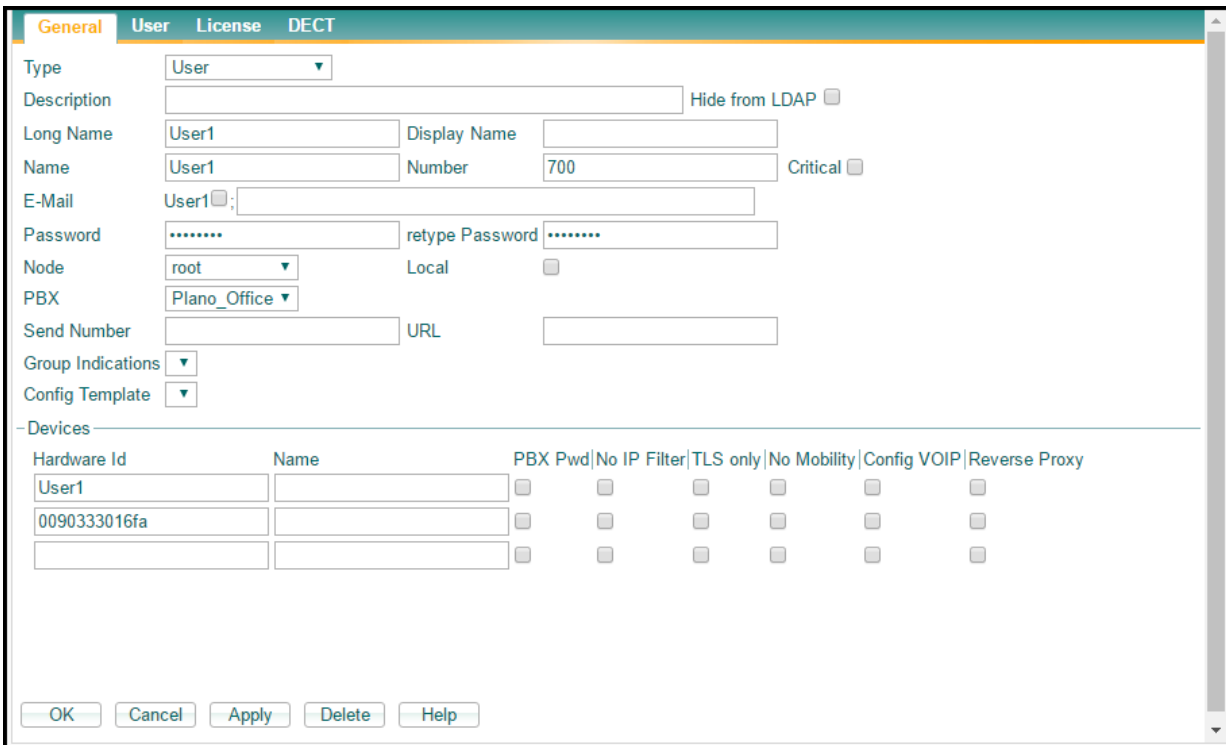
The Provider will send and receive all numbers (CGPN & CDPN) in international number format.

<b>CGPN In</b>			
<input type="text"/>	<input type="text" value="0"/>	→	<input type="text" value="0"/>
<input type="text"/>	<input type="text"/>	→	<input type="text" value="00"/>
<input type="text"/>	<input type="text"/>	→	<input type="text"/>
<b>CDPN In</b>			
<input type="text" value="International"/>	<input type="text" value="41438198"/>	→	<input type="text"/>
<input type="text"/>	<input type="text" value="0438198"/>	→	<input type="text"/>
<input type="text"/>	<input type="text"/>	→	<input type="text"/>
<b>CGPN Out</b>			
<input type="text" value="900"/>	→	<input type="text" value="International"/>	<input type="text"/>
<input type="text"/>	→	<input type="text" value="International"/>	<input type="text" value="41438198"/>
<input type="text"/>	→	<input type="text"/>	<input type="text"/>
<b>CDPN Out</b>			
<input type="text" value="00"/>	→	<input type="text" value="International"/>	<input type="text"/>
<input type="text" value="0"/>	→	<input type="text" value="International"/>	<input type="text" value="41"/>
<input type="text"/>	→	<input type="text" value="International"/>	<input type="text" value="41"/>
<input type="text"/>	→	<input type="text"/>	<input type="text"/>

## 5.2 Configuring SIP Extension

Below is the configuration of a SIP extension on the PBX used for the test.

1. Login into IPVA → PBX → Objects
2. To create a new extension, Select **User** from the drop down menu.
3. Add a new user by clicking the **NEW** button
4. Select the **Name** and **Number**.
5. Provide the mac address and the user id (User1) of the phone as shown below and clock **Apply**.



**General** | User | License | DECT

Type: User ▼

Description:  Hide from LDAP ☐

Long Name: User1 Display Name:

Name: User1 Number: 700 Critical ☐

E-Mail: User1@

Password:  retype Password:

Node: root ▼ Local: ☐

PBX: Plano\_Office ▼

Send Number:  URL:

Group Indications: ▼

Config Template: ▼

— Devices —

Hardware Id	Name	PBX Pwd	No IP Filter	TLS only	No Mobility	Config VOIP	Reverse Proxy
User1	<input type="text"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
0090333016fa	<input type="text"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="text"/>	<input type="text"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

OK Cancel Apply Delete Help



### 5.3 Configuring FAX Extension

Below is the configuration of a SIP extension on the PBX used for the test.

1. Login into IPVA → PBX → Objects
2. To create a new extension, Select **User** from the drop down menu.
3. Add a new user by clicking the **NEW** button
4. Select the **Name** and **Number**.
5. Provide the mac address and the user id (User3) of the phone as shown below **Apply**.

General User License DECT

Type

User

Description

Hide from LDAP

Long Name

User3

Display Name

Name

User3

Number

703

Critical

E-Mail

User3

Password

.....

retype Password

.....

Node

root

Local

PBX

Plano\_Office

Send Number

URL

Group Indications

Config Template

Devices

Hardware Id	Name	PBX Pwd	No IP Filter	TLS only	No Mobility	Config VOIP	Reverse Proxy
User3							
0090331C1F22							

OK

Cancel

Apply

Delete

Help