

Yeastar P-Series PBX Appliance-Edition

SIP Trunk PBX Configuration Guide for
Swisscom Smart Business Connect





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1 Introduction

1.1 Objective and purpose

Describes the SIP Trunk configuration of IP PBX or communication systems. The IP PBX or communications systems are homologated using this SIP Trunk configuration in order to interoperate with Swisscom service which using demarcation with eSBC or IMG at customer premise.

1.2 Target audience

IP PBX and Communication System Integrators, who have joined the Swisscom Partner Training for Connect Trunk.

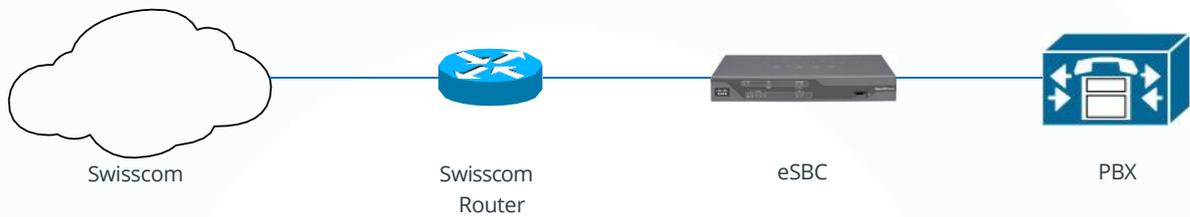
1.3 Terms, abbreviations

Terms	Abbreviations
SIP	Session Initiation Protocol
IP	Internet Protocol
PBX	Private Branch Exchange
eSBC	Enterprise Session Border Controller
IMG	ISDN Media Gateway



2 Overview PBX

2.1 SIP Trunk network architecture customer side



2.2 Hardware requirements

Yeastar P-Series Appliance-Edition P520/P550/P560/P570, and Yeastar P-Software-Edition

2.3 Software requirements

Yeastar P-Series Appliance-Edition P5X SW37.7.0.51 and later

2.4 Support Contacts

Yeastar

helpdesk@yeastar.com

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3 SIP Trunk features

3.1 Features supported and tested

- National calls
- International calls
- National calls with international prefix
- Toll free numbers (0800, 0900)
- DTMF(RFC 2833)
- Call cancellation
- Call rejection (see remark 3.2)
- Calls with early media
- Calls to special/short numbers
- Calling line indication presentation (CLIP)
- Calling line indication restriction (CLIR) (see remark 3.2)
- Special Arrangement
- Call hold/resume
- Music on hold
- Call forwarding unconditional (see remark 3.2)
- Call forwarding busy (see remark 3.2)
- Call forwarding no answer (see remark 3.2)
- Attended call transfer
- Blind call transfer
- 3-party conference
- Fax to e-Mail
- Fax FXS Port PBX T.38 (max. BitRate 4'800bit/s).

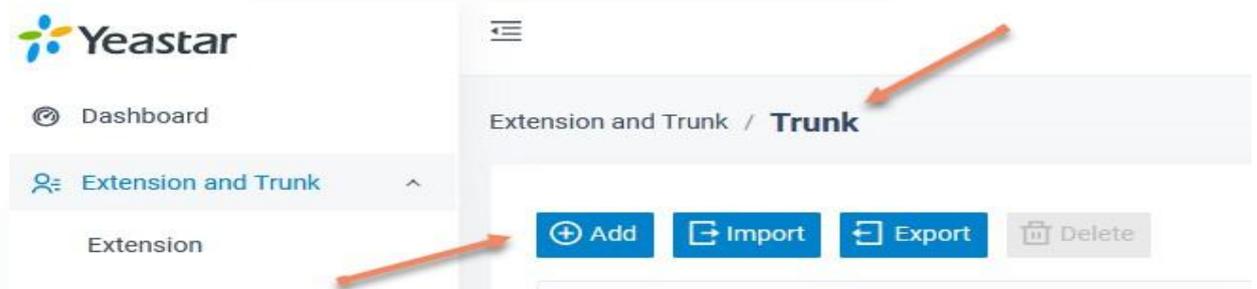
3.2 Caveats and known restrictions

- Call with CLIR only with additional outbound route, and outbound caller ID "anonymous". In the case of external call diversion, and CLIR from user A, the CLIP from user B (PBX) is displayed to user C.
- Fax, ideally with an FXS port on the PBX, or incoming FAX to E-Mail, and the T.38 settings in the PBX! (Fax with a SIP-a/b adapter, not all FAX connections are guaranteed. Fax G711 also works only partially and is therefore not recommended)!
- Call rejected: the user sends 486 "User Busy" Various rejection destinations can be set in the PBX: Voice mail, other destination, etc.
- Call diversion: is always in the PBX with the establishment of a second connection. Call forwarding 302 is not possible!
- Modem not tested.
- Billing with Special Arrangement: the billing will be done on the trunk main number instead of the user number.



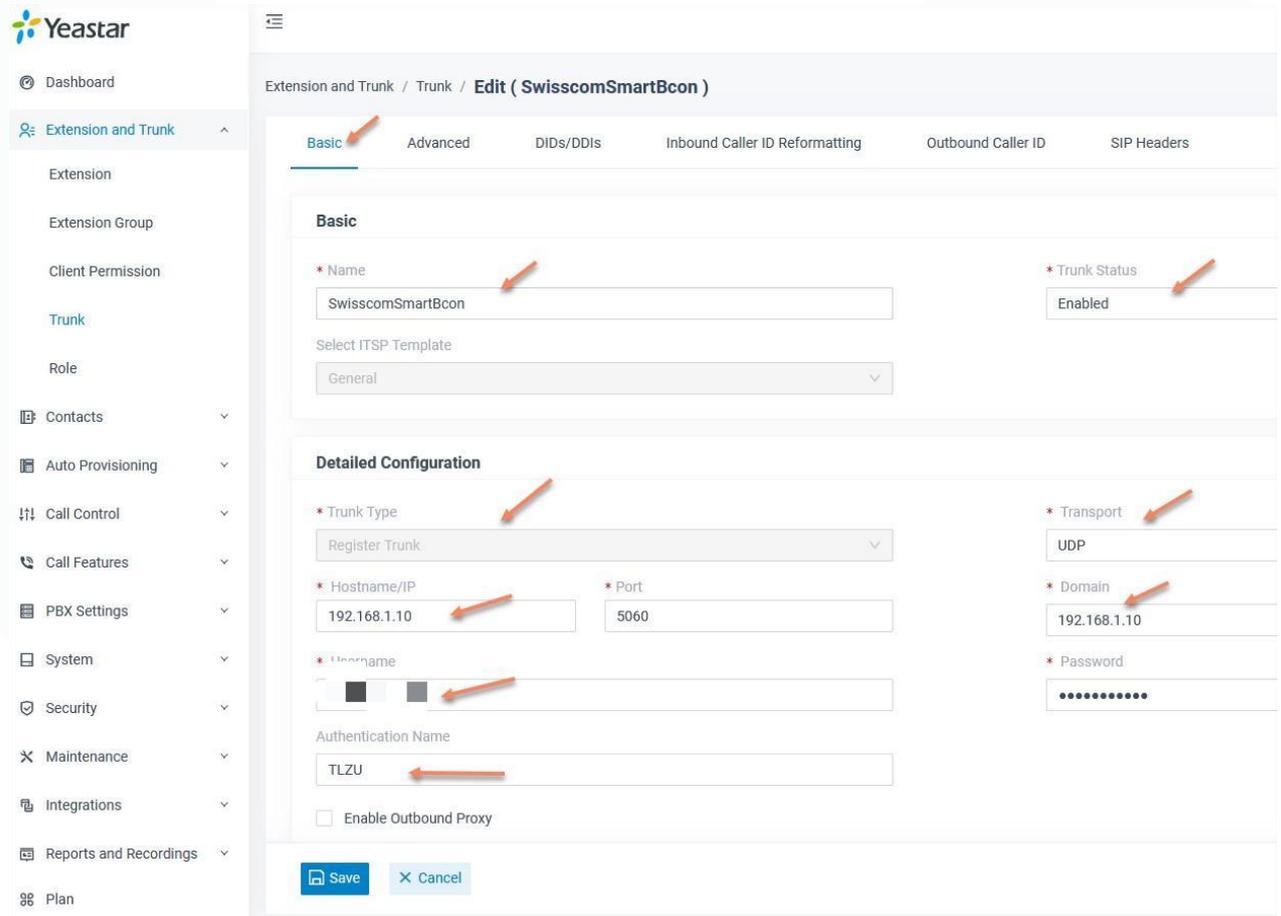
4 SIP Trunk configuration PBX

side Extension and Trunk/Trunk



Extension and Trunk/Trunk/Edit (SwisscomSmartBcon) Basis

Note: There will be a template later (Select ITSP Template)





Extension and Trunk/Trunk/Edit (Swisscom) Advanced

The screenshot shows the Yeastar web interface for configuring a SIP Trunk. The left sidebar contains a navigation menu with categories like Extension and Trunk, Trunk, and Role. The main content area is titled "Extension and Trunk / Trunk / Edit (SwisscomSmartBcon)". The "Advanced" tab is selected, and the "Codec Settings" section is visible. It features two columns: "Available" (11 items) and "Selected" (3 items). The "Available" list includes Codec, GSM, H264, H263, H263P, G722, G726, and G729. The "Selected" list includes Codec, a-law, u-law, and iLBC. Below the codec lists are "VoIP Settings" for "DTMF Mode" (RFC4733 (RFC2833)) and "DTMF fmp" (0-16). "Save" and "Cancel" buttons are at the bottom.

This screenshot shows the "VoIP Settings" and "Call Restriction" sections of the configuration page. The "VoIP Settings" section includes "DTMF Mode" (RFC4733 (RFC2833)) and "DTMF fmp" (0-16). Below these are several checkboxes: "Qualify" (checked), "Enable SRTP" (unchecked), "T:38 Support" (checked), "Inband Progress" (checked), and "Ignore 183 Message without SDP" (checked). The "Call Restriction" section includes "Call Restriction Type" (Outbound Call) and "Maximum Concurrent Calls" (Unlimited). "Save" and "Cancel" buttons are at the bottom.



Extension and Trunk/Trunk/Edit (Swisscom) DID/DDIs

The screenshot shows the Yeastar web interface for configuring a SIP Trunk. The left sidebar contains a navigation menu with categories like Dashboard, Extension and Trunk, Client Permission, Trunk, Role, Contacts, Auto Provisioning, Call Control, Call Features, PBX Settings, System, Security, Maintenance, Integrations, Reports and Recordings, and Plan. The main content area is titled "Extension and Trunk / Trunk / Edit (SwisscomSmartBcon)". Below the title are tabs for "Basic", "Advanced", "DIDs/DDIs", "Inbound Caller ID Reformatting", "Outbound Caller ID", and "SIP Headers". The "DIDs/DDIs" tab is active, showing a table with columns for "DID/DDI", "DID/DDI Name", "Move", and "Operations". The table contains 10 rows of data, each with a checkbox, a DID/DDI value, a name, and move/edit/delete icons. At the bottom of the interface are "Save" and "Cancel" buttons. Two red arrows point to the "DIDs/DDIs" tab and the "Save" button.

Yeastar

Dashboard

Extension and Trunk

Extension

Extension Group

Client Permission

Trunk

Role

Contacts

Auto Provisioning

Call Control

Call Features

PBX Settings

System

Security

Maintenance

Integrations

Reports and Recordings

Plan

Extension and Trunk / Trunk / Edit (SwisscomSmartBcon)

Basic Advanced **DIDs/DDIs** Inbound Caller ID Reformatting Outbound Caller ID SIP Headers

+ Add Import Export Delete

DID/DDI	DID/DDI Name	Move	Operations
<input type="checkbox"/>		⬅️ ⬆️ ⬇️ ⬇️	
<input type="checkbox"/>		⬅️ ⬆️ ⬇️ ⬇️	
<input type="checkbox"/> 4		⬅️ ⬆️ ⬇️ ⬇️	
<input type="checkbox"/> 4		⬅️ ⬆️ ⬇️ ⬇️	
<input type="checkbox"/>		⬅️ ⬆️ ⬇️ ⬇️	
<input type="checkbox"/>		⬅️ ⬆️ ⬇️ ⬇️	
<input type="checkbox"/>		⬅️ ⬆️ ⬇️ ⬇️	
<input type="checkbox"/> 4		⬅️ ⬆️ ⬇️ ⬇️	
<input type="checkbox"/> 4		⬅️ ⬆️ ⬇️ ⬇️	
<input type="checkbox"/> 4		⬅️ ⬆️ ⬇️ ⬇️	

Save Cancel



Extension and Trunk/Trunk/Edit (Swisscom) Inbound Caller ID Reformatting

The screenshot shows the Yeastar web interface for configuring Inbound Caller ID Reformatting. The breadcrumb path is "Extension and Trunk / Trunk / Edit (SwisscomSmartBcon)". The "Inbound Caller ID Reformatting" tab is selected. The interface includes a table with columns for Patterns, Strip, Prepend, Move, and Operations. Two rows are visible: one for "+41" with Strip set to 3 and Prepend set to 0, and another for "+" with Strip set to 1 and Prepend set to 00. A "Save" button is at the bottom left.

Patterns	Strip	Prepend	Move	Operations
<input type="checkbox"/> +41	3	0	⌵ ⌶ ⌷ ⌸ ⌹	
<input type="checkbox"/> +	1	00	⌵ ⌶ ⌷ ⌸ ⌹	

Extension and Trunk/Trunk/Edit (Swisscom) Outbound Caller ID

The screenshot shows the Yeastar web interface for configuring Outbound Caller ID. The breadcrumb path is "Extension and Trunk / Trunk / Edit (Swisscom)". The "Outbound Caller ID" tab is selected. The "General" section has two input fields: "Outbound Caller ID" and "Outbound Caller ID Name". The "Outbound Caller ID List" section includes a table with columns for Outbound Caller ID, Outbound Caller ID Name, Associated Extensions, Move, and Operations. One row is visible with "0442747671-0442747677" and "1001-1007".

Outbound Caller ID	Outbound Caller ID Name	Associated Extensions	Move	Operations
<input type="checkbox"/> 0442747671-0442747677		1001-1007	⌵ ⌶ ⌷ ⌸ ⌹	



Extension and Trunk/Trunk/Edit (Swisscom) SIP Headers

Extension and Trunk / Trunk / Edit (SwisscomSmartBcon)

Basic Advanced DIDs/DDIs Inbound Caller ID Reformatting Outbound Caller ID **SIP Headers** ←

Inbound Parameters

* Get Caller ID From: From ↗

* Get DID From: To ↗

Outbound Parameters

* From User Part: Originator Caller ID ↗

* From Display Name Part: [Default]

Diversion: [Default]

Remote-Party-ID: [None]

P-Asserted-Identity: [None]

P-Preferred-Identity: [None]

Other Settings

User Agent:

Realm:

Send Privacy ID

Extension and Trunk / Trunk / Edit (SwisscomSmartBcon)

Basic Advanced DIDs/DDIs Inbound Caller ID Reformatting Outbound Caller ID **SIP Headers** ←

Originator caller ID:

Diversion: [Default]

Remote-Party-ID: [None]

P-Asserted-Identity: [None]

P-Preferred-Identity: [None]

Other Settings

User Agent:

Realm:

Send Privacy ID

User Phone

100rel

* Maxptime: [Default]

Support P-Early-Media



Call Control/Inbound Route

Call Control / **Inbound Route**

[Add](#) [Import](#) [Export](#) [Delete](#)

<input type="checkbox"/>	Name	DID Patterns	Caller ID Pattern	Default Destination	Current Destination	Time-based Routing Mode	Move	Operations
<input type="checkbox"/>	DDI70	+4144274767...		IVR	IVR	Disabled	↕ ^	Edit Delete
<input type="checkbox"/>	DDI71-74	+41442747671~+41442747674		Match DID Range to Extension Range	Match Extension Range	Disabled	↕ ^	Edit Delete
<input type="checkbox"/>	DDI75_FAX_EMAIL	+4144274767...		Fax To Email	Fax To Email	Disabled	↕ ^	Edit Delete
<input type="checkbox"/>	DDI76_FAX_Out_In	+4144274767...		Extension	Extension	Disabled	↕ ^	Edit Delete

Total: 4 20 / page

Call Control/Outbound Route

Call Control / **Outbound Route**

[Add](#) [Import](#) [Export](#) [Delete](#)

<input type="checkbox"/>	Name	Outbound Caller ID	Dial Pattern	Trunk	Extension/Group	Move	Operations
<input type="checkbox"/>	SwisscomS...	Anonymous	7X	SwisscomS...	1001-1001 App1 1002-1002 App2 1003-1003 App3 ...	↕ ^ v ↓	Edit Delete
<input type="checkbox"/>	SwisscomSBC		X	SwisscomS...	1001-1001 App1 1002-1002 App2 1003-1003 App3 ...	↕ ^ v ↓	Edit Delete

this outbound route is only necessary if there is an option to send "anonymous".

normal outbound route



PBX Settings/SIP Settings/T.38

PBX Settings / **SIP Settings**

General Codec TLS Session Timer QoS **T.38** Advanced

T.38

* T.38 Max BitRate

4800 

No T.38 Attributes in re-INVITE SDP

Error Correction Mode



5 SIP Trunk recommendations

5.1 DTMF

5.1.1 Sending (to Swisscom)

DTMF Signals SHALL be sent according to the IETF RFC's 2833/4733. SIP INFO is currently NOT supported. In cases where DTMF Tones are sent in-band in a G.711 RTP Stream, it is transparent to the Network and proper DTMF transmission across the Network can therefore not guarantee by either side.

In case of DTMF transmission the SDP MUST contain the *rtpmap* and *fmtp* attributes associated with the DTMF payload.

Swisscom recommendation:

DTMF Signals sent according to the IETF RFC's 2833/4733 offer the best compatibility with most systems.

5.1.2 Receiving (from Swisscom)

To insure (backward) compatibility with system who do not support/send out-of-band DTMF (RFC 2833), a system MUST be capable to accept both, in-band DTMF (G.711 payload) and out-of-band DTMF (RFC's 2833/4733)

Swisscom recommendation:

In cases the system is depending on DTMF Signals (e.g. Contact Center, Voicemail, etc.) and is not capable to handle both DTMF methods, it is in the responsibility of the solution provider to install appropriate equipment to convert between the two signaling methods

5.2 Best Practices

5.2.1 Fax over IP Recommendations & Settings

For Fax Transmissions please read our published recommendation and white papers.

Swisscom Recommendation:

[Fax over Smart BCon](#)