Yeastar P-Series PBX Appliance-Edition

SIP Trunk PBX Configuration Guide for Swisscom Smart Business Connect

From: Alltron AG (cover by Yeastar)

Date: 03.03.2025

Subject: SIP Trunk PBX Configuration Guide for Swisscom

Scope	IP PBX and communication system homologation with Swisscom
Doc. ID	Swisscom_Trunk_Configuration_Guide_PBX_Template_v1.3-en.blank.docx
Version	1.0
Status	Available
Date of issue	03.03.2025
Valid from	Approval by Product Manager
Valid until	New version available
Document name	SIP_Trunk_PBX_Configuration_Guide_for_Swisscom_Smart_Business_Connect_en.pdf

Contents

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

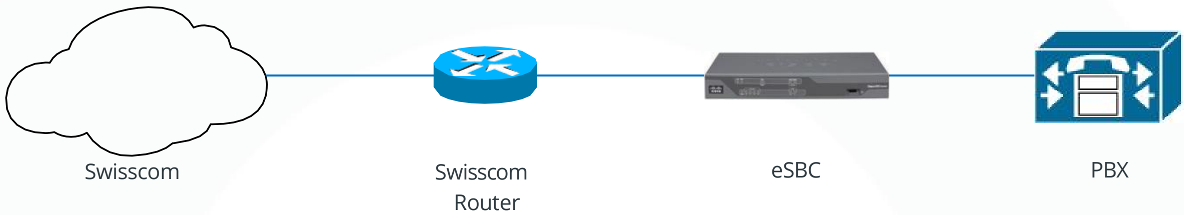
IP PBX and Communication System Integrators, who have joined the Swisscom Partner Training for Connect Trunk.

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

Alltron AG
Rolf Käfer
Hintermättlistrasse
3 CH-5506
Mägenwil

062 889 61 39
rolf.kaefer@alltron.ch

3 SIP Trunk features

3.1 Features supported and tested

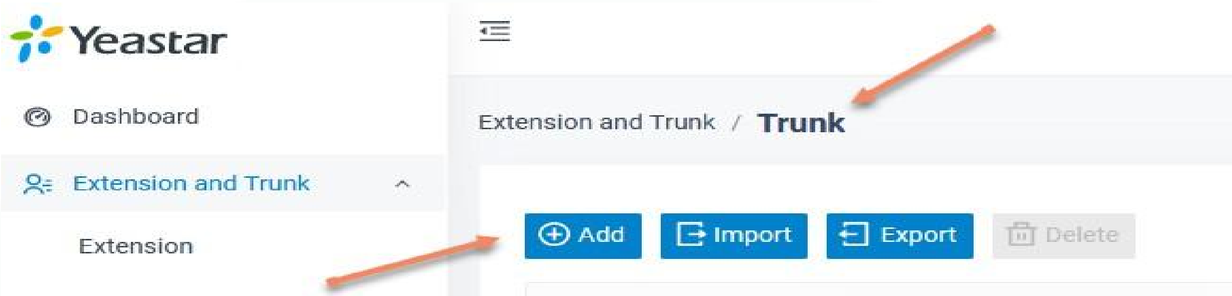
- National calls
- International calls
- National calls with international prefix
- Toll free numbers (0800, 0900)
- DTMF (RFC 2833)
- Call cancellation
- Call rejection (see remark 3.2)
- Calls with early media
- Calls to special/short numbers
- Calling line indication presentation (CLIP)
- Calling 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.

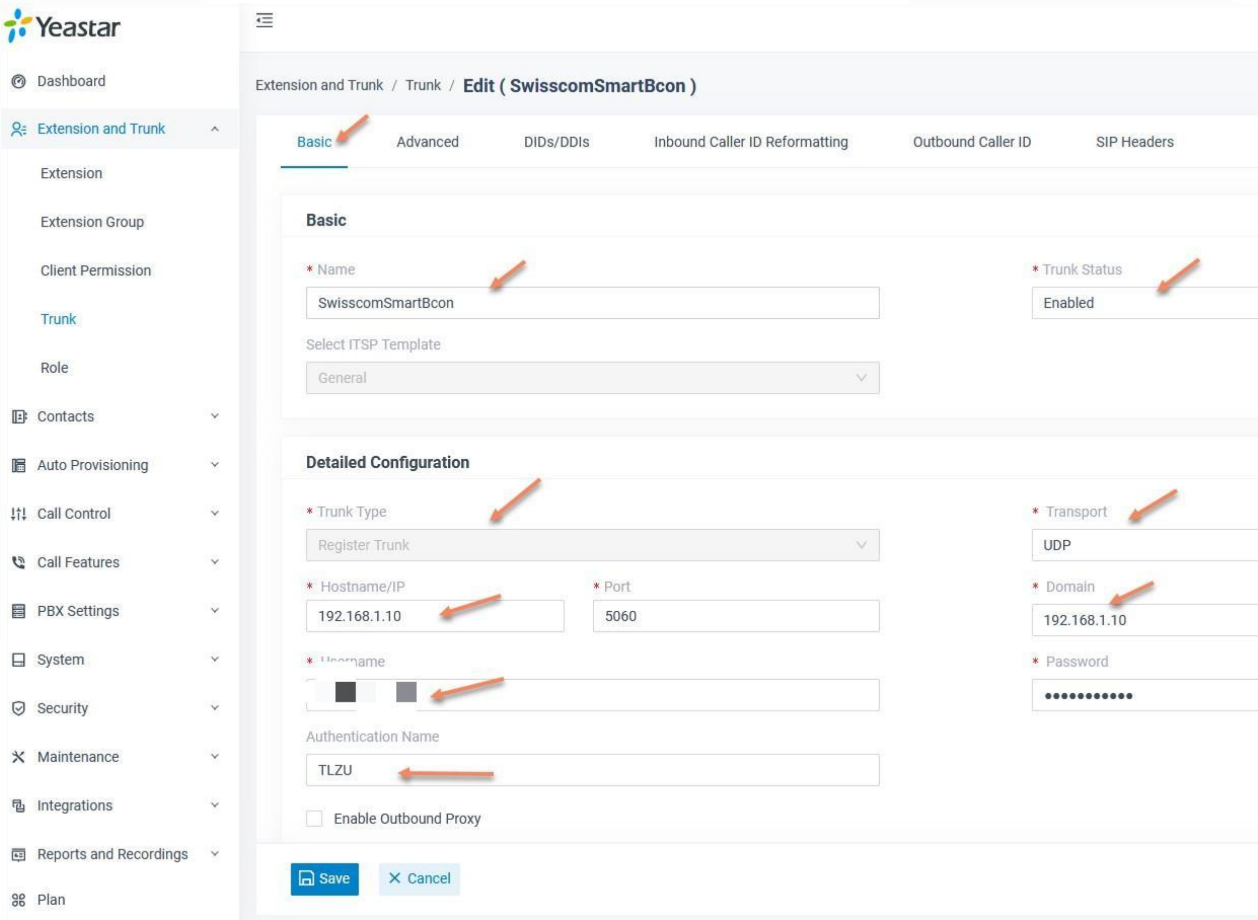
4 SIP Trunk configuration PBX

side Extension and Trunk/Trunk

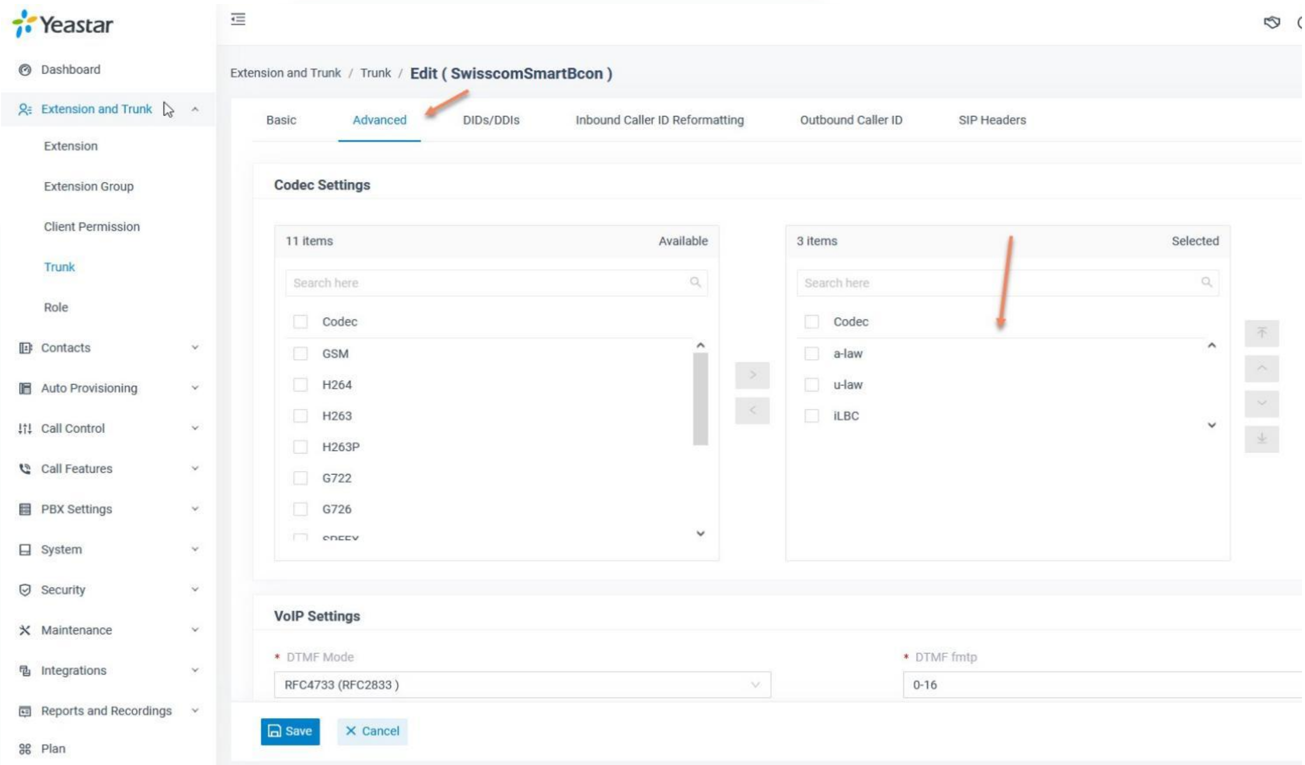


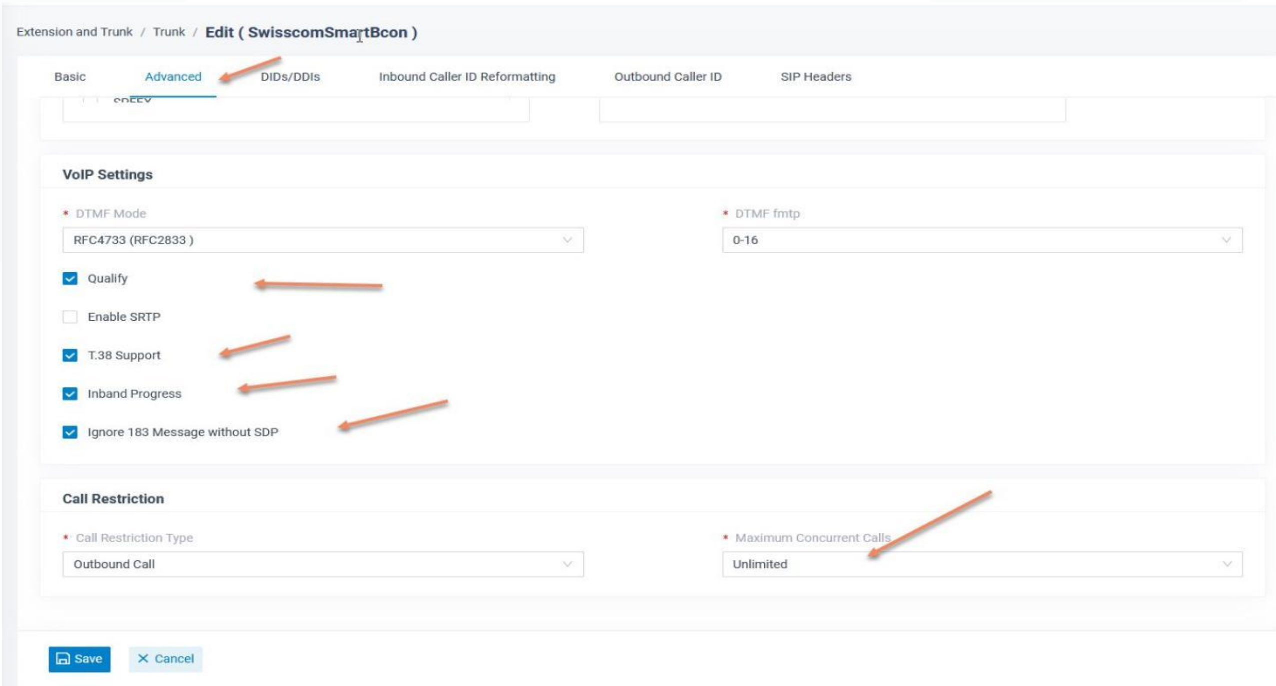
Extension and Trunk/Trunk/Edit (SwisscomSmartBcon) Basis

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

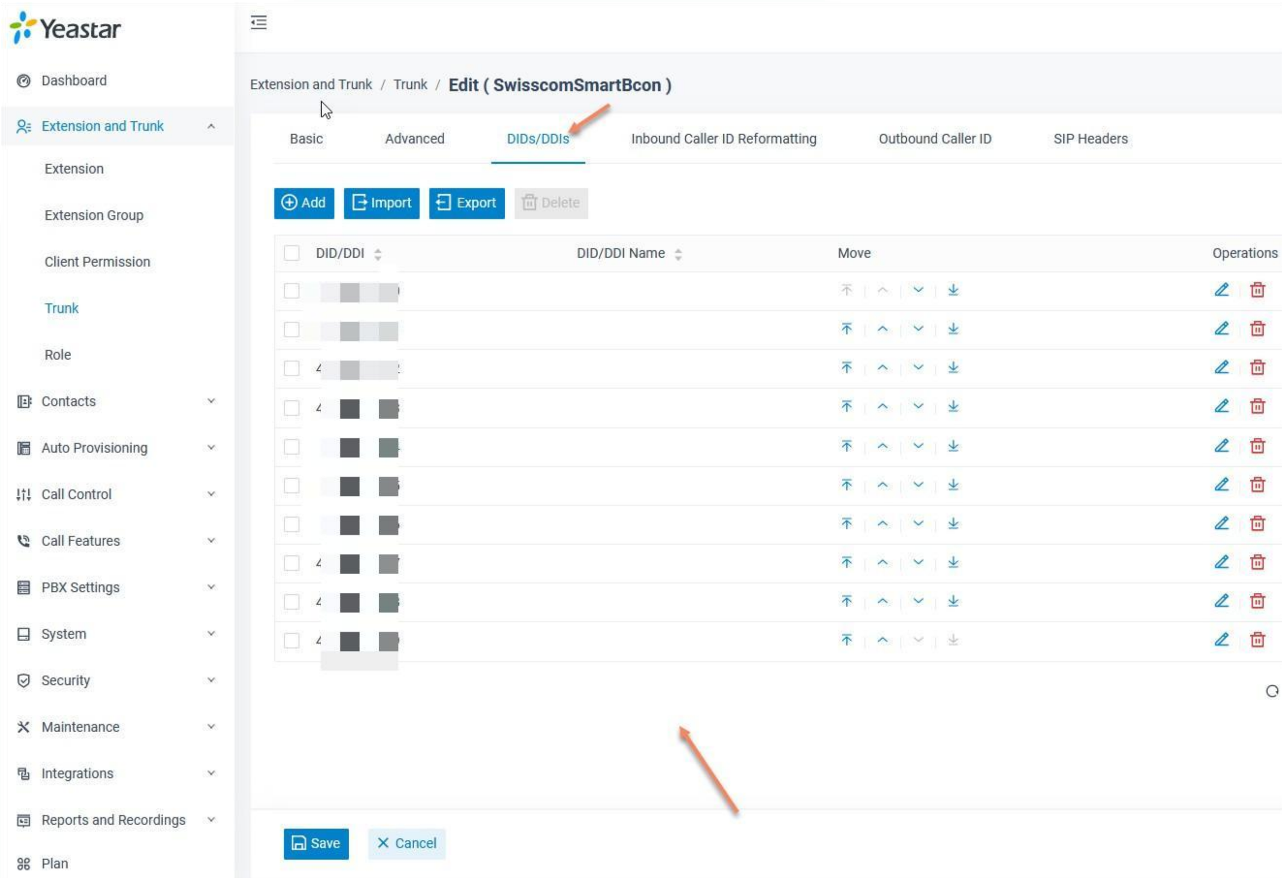


Extension and Trunk/Trunk/Edit (Swisscom) Advanced

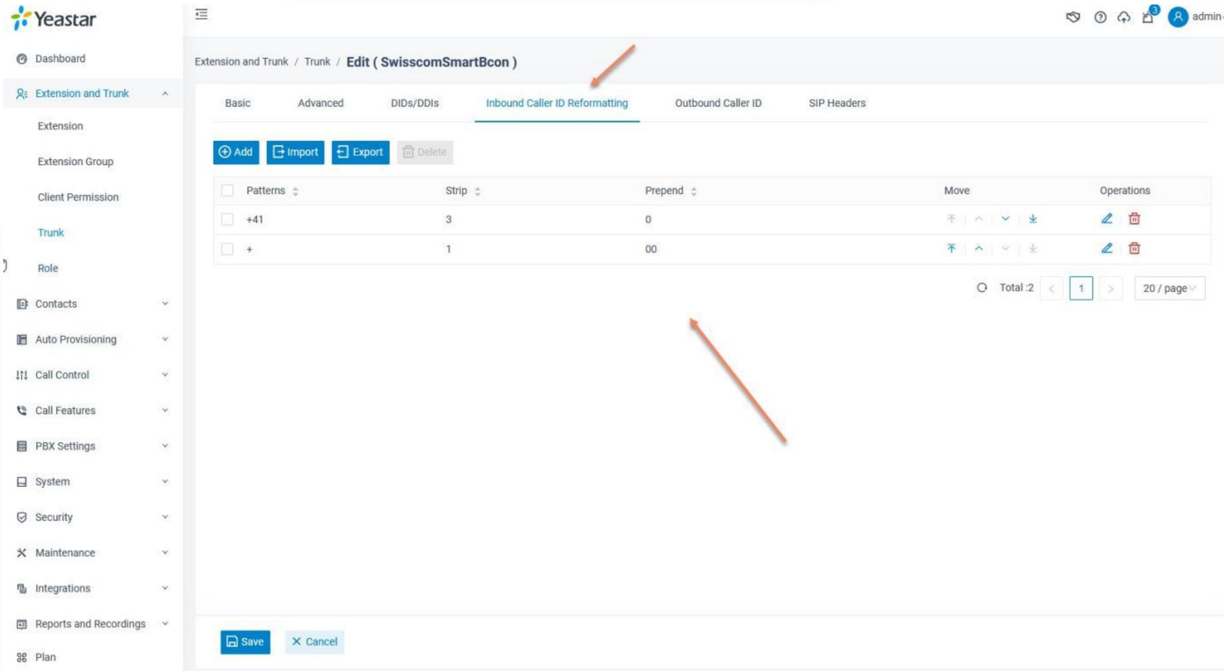




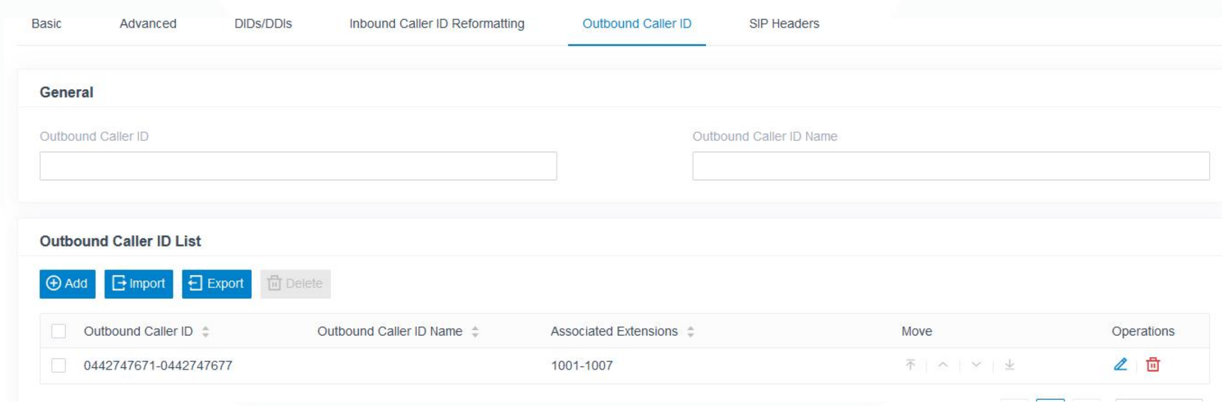
Extension and Trunk/Trunk/Edit (Swisscom) DID/DDIs



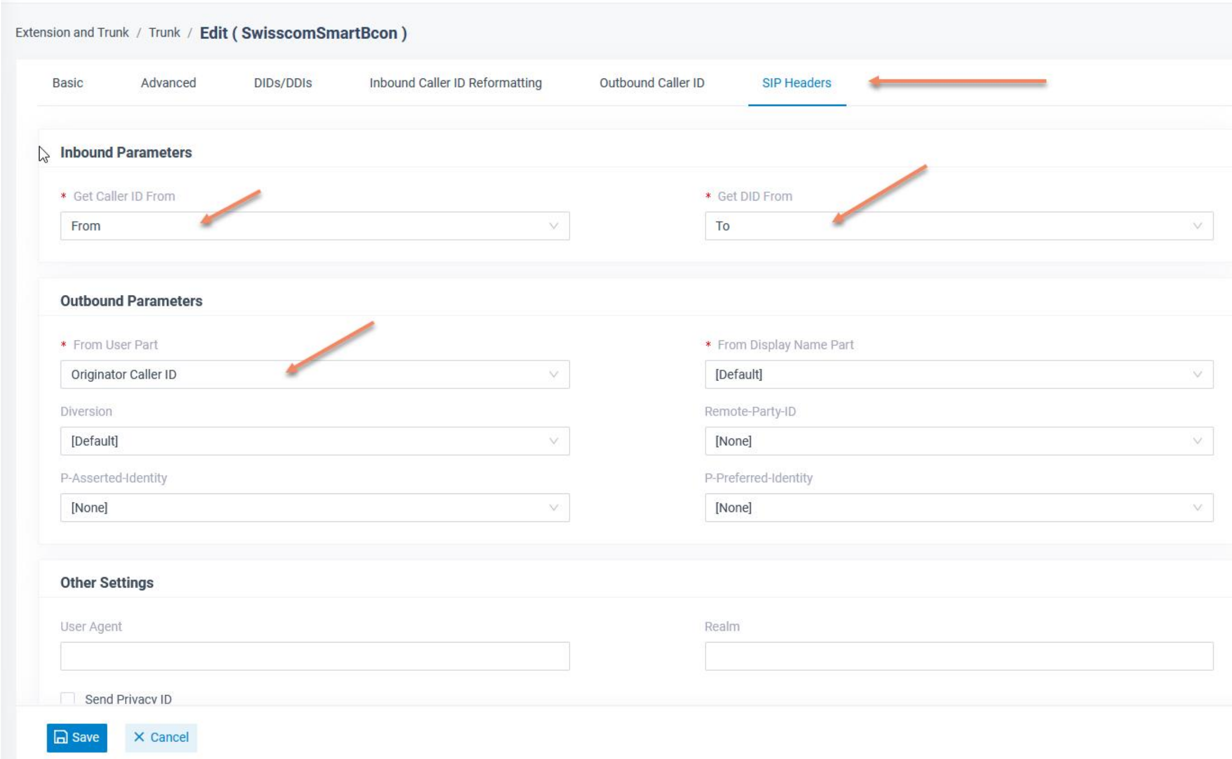
Extension and Trunk/Trunk/Edit (Swisscom) Inbound Caller ID Reformatting

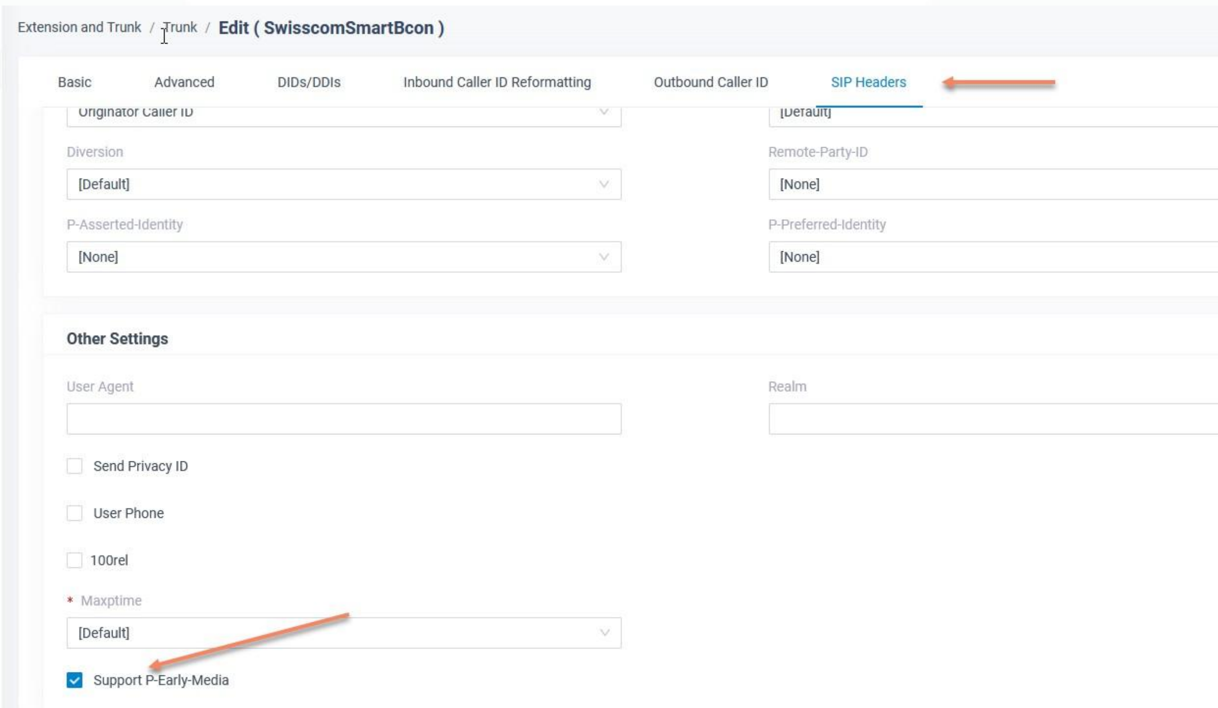


Extension and Trunk/Trunk/Edit (Swisscom) Outbound Caller ID

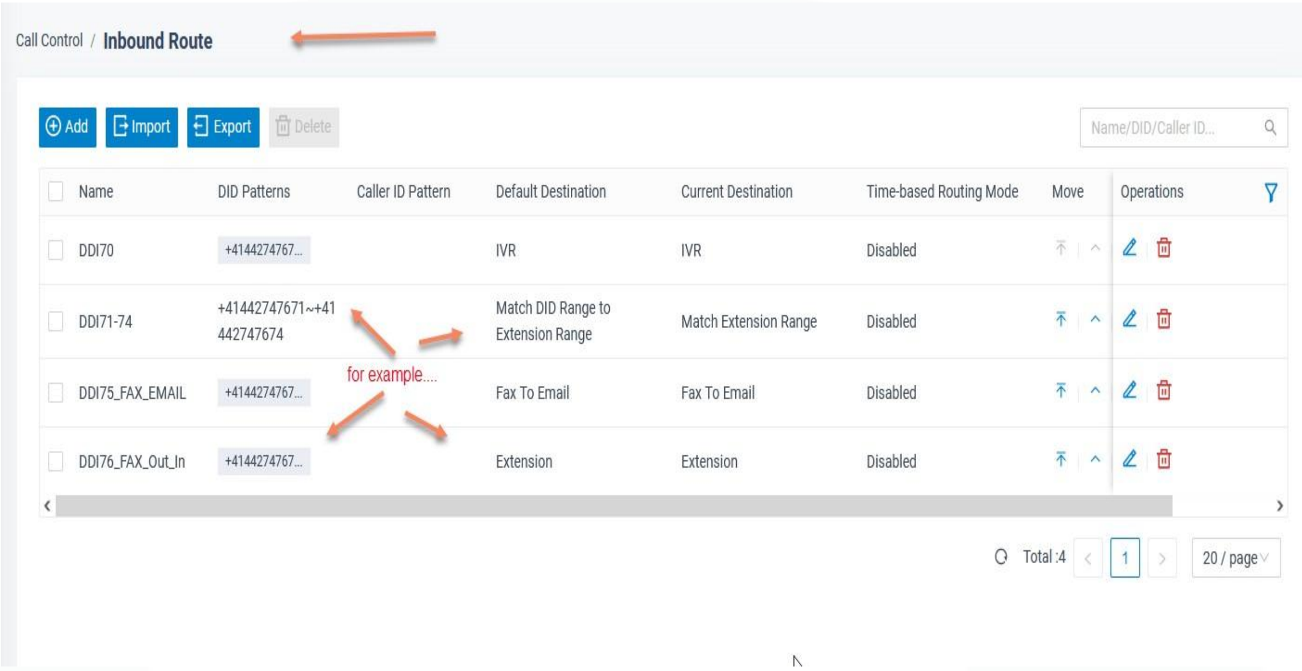


Extension and Trunk/Trunk/Edit (Swisscom) SIP Headers

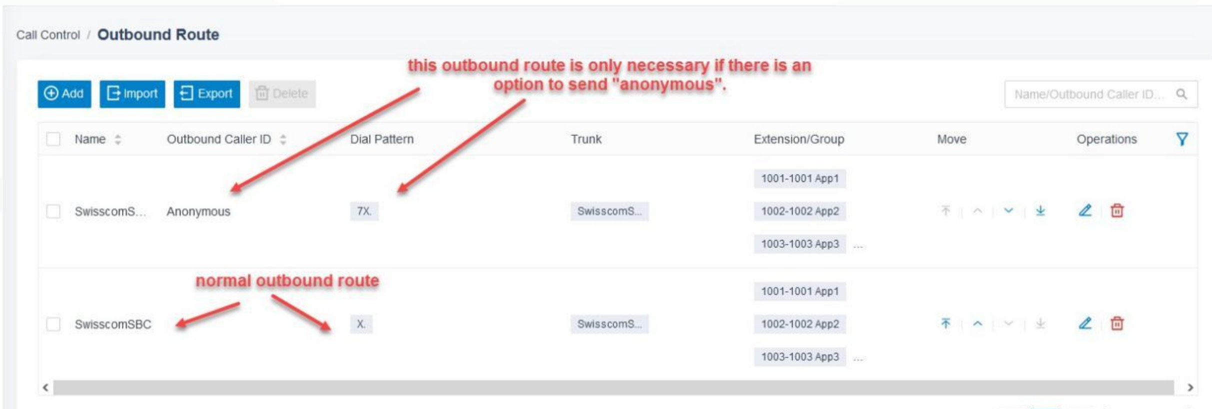




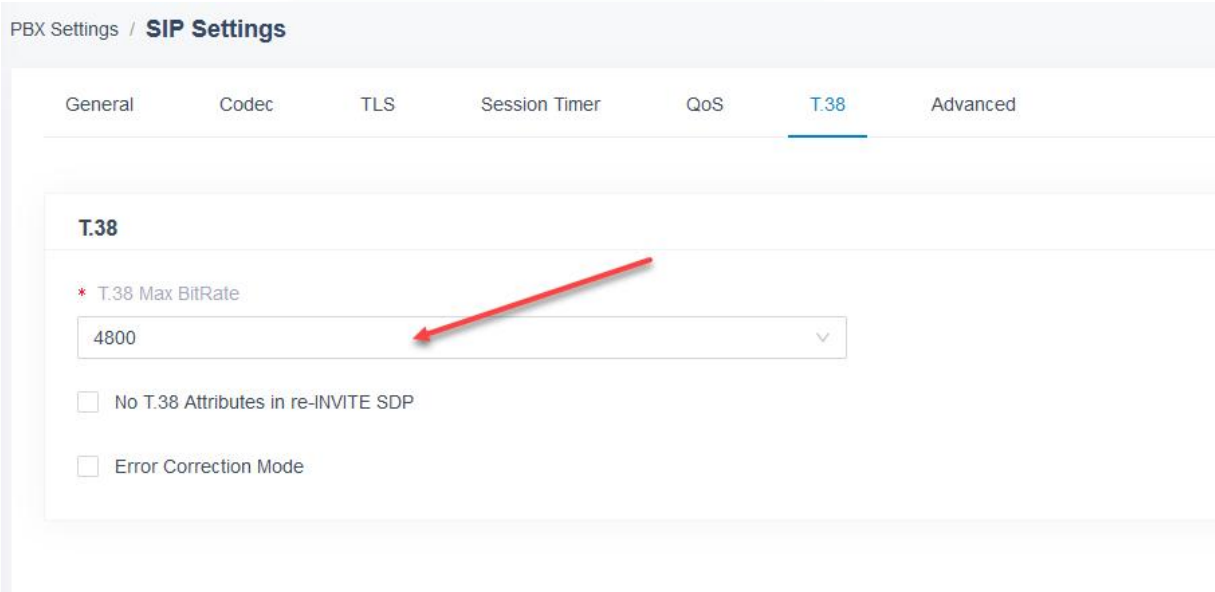
Call Control/Inbound Route



Call Control/Outbound Route



PBX Settings/SIP Settings /T.38



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