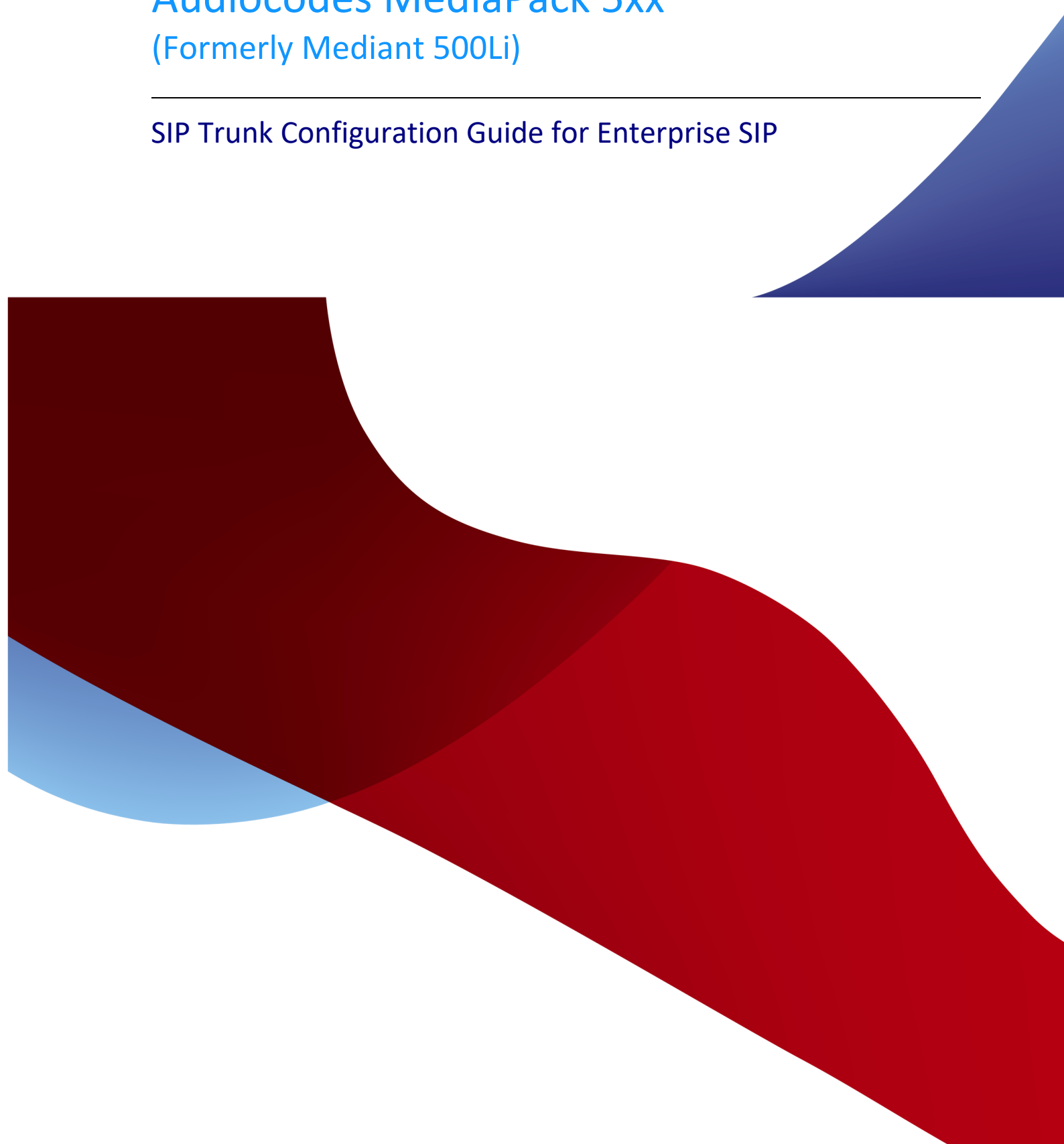


# Audiocodes MediaPack 5xx

(Formerly Mediant 500Li)

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SIP Trunk Configuration Guide for Enterprise SIP



**From** Enterprise SIP Engineer  
**Date**  
**Subject** SIP Trunk Configuration Guide for Enterprise SIP

**To** IP PBX and  
Communication System  
Integrators  
**For information**

|                         |                                                                          |
|-------------------------|--------------------------------------------------------------------------|
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### Change history

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| 0.1     | 05.07.2023 | Norbert Moura  | Original configuration guide |
|         |            |                |                              |
|         |            |                |                              |
|         |            |                |                              |

### Review

| Version | Review date | Reviewed by | Comments |
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|         |             |             |          |
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|         |             |             |          |

### Release

| Version | Date of release | Released by   | Comments |
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| 1.0     | 18.08.2023      | Norbert Moura |          |
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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 the Enterprise SIP service.

More information about the Enterprise SIP service is available here:  
<https://www.swisscom.ch/de/business/enterprise/angebot/wireline-networks-bundles/business-voice/sip-trunking.html>

### 1.2 Target audience, requirements for the reader

IP PBX and Communication System Integrators.

### 1.3 Terms, abbreviations

| Terms | Abbreviations                                                                                                                     |
|-------|-----------------------------------------------------------------------------------------------------------------------------------|
| ATA   | Analog Telephone Adapter, used to integrate analog endpoints into VoIP systems                                                    |
| eSBC  | SBC build into the Swisscom Cisco router                                                                                          |
| LIS   | Location Information Service of Skype for Business                                                                                |
| PSTN  | Public Switched Telephone Network                                                                                                 |
| SBC   | Session Boarder Controller, device for termination of voice over IP networks. It controls signalization as well as media streams. |

### 1.4 Referenced documents

- [1] SIP Trunk Technical User Network Interface Guide for Enterprise SIP (UNI Guide)

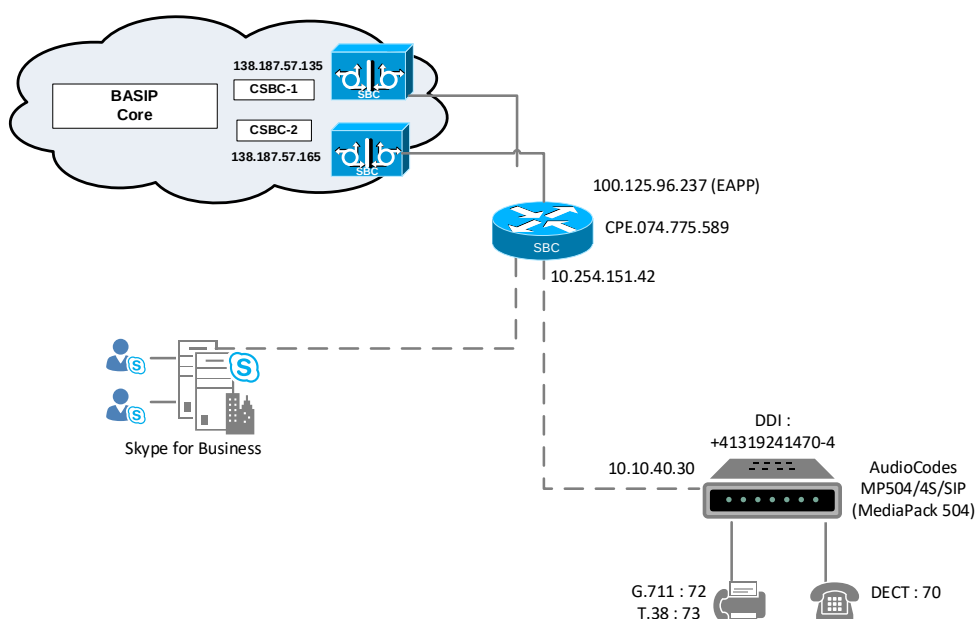
## 2 Overview

This document describes only the configuration parameters for Audiocodes MediaPack 5xx (formerly Audiocodes Mediant 500L-i, rebranding done by Mid of April 2023)

Only the Standard Enterprise SIP trunk (without redundancy) is described in detail. For high availability and disaster recovery scenario please contact Swisscom Spec Sales and Design team over your account manager to design an appropriate system architecture to match the customers' need.

### 2.1 SIP Trunk network architecture

The Enterprise SIP Trunk is terminated on the eSBC, located on of the Cisco router.



### 2.2 Hardware requirements

This Audiocodes device can be used in conjunction with Skype for Business Server 2015 and Lync Server 2013. Please take the configuration's parameters for Skype for Business Server 2015 and Lync Server 2013 from the respective homologations.

### **2.3 Software requirements**

The homologation was made with this version :

- Audiocodes MediaPack 504 (MP504) version 7.26A.356.174

**Note:**

Older product version and builds were not tested and may not work properly

### **3 SIP Trunk features**

#### **3.1 Features supported and tested**

- National calls
- International calls
- National calls with international prefix
- Toll free numbers (0800)
- DTMF (RFC 2833, default Payload Type 101)
- Call cancellation
- Call rejection
- Calls with early media
- Calls to special/short numbers
- Calling line indication presentation (CLIP)
- Calling line indication restriction (CLIR)
- Special Arrangement
- Call hold/resume
- Music on hold
- Call forwarding unconditional
- Call forwarding busy
- Call forwarding no answer
- Attended call transfer
- Blind call transfer
- 3-party conference
- T.38 Fax
- Fax with G.711 pass-through
- Keep alive out of session

#### **3.2 Caveats and known restrictions**

- Modem not tested
- Billing with Special Arrangement: the billing will be done on the trunk main number instead of the user number

##### **3.2.1 Features not supported**

- E112 with geolocation. Instead, the geolocation is based on A-number and performed in SIP Core.



#### 4 SIP Trunk configuration Audiocodes MediaPack 5xx (MP5xx)

The below settings are made upon the factory reset of the box. They ensure that the full test plan will be successful. As soon as a setting differs from the default, it is visible as below:



The IP network part as well as the core entities of the Signaling & Media are not described in the current document.

For Basic configuration please refer to the [MediaPack 5xx Voice Gateway Basic Configuration](#) document.

##### 4.1 General settings

Signaling & Media->SIP Definitions->Message Structure

| Parameter              | Value                   |
|------------------------|-------------------------|
| Asserted Identity Mode | Add P-Asserted Identity |

Signaling & Media->SIP Definitions->Transport Settings

| Parameter          | Value |
|--------------------|-------|
| SIP Transport Type | TCP   |

Signaling & Media->SIP Definitions->Proxy & Registration

| Parameter                    | Value  |
|------------------------------|--------|
| Use Gateway Name for OPTIONS | Server |
| Registrar Transport Type     | TCP    |
| Always Use Proxy             | Enable |

Signaling & Media->Gateway->DTMF & Supplementary->DTMF & Dialing

| Parameter               | Value |
|-------------------------|-------|
| Max Digits In Phone Num | 25    |

Signaling & Media->MediaRTP/RTCP Settings

| Parameter                            | Value   |
|--------------------------------------|---------|
| Comfort Noise Generation Negotiation | Disable |

Signaling & Media->Message Manipulation->Message Manipulations

| Manipulation Set ID 0 |                                  |
|-----------------------|----------------------------------|
| Parameter             | Value                            |
| Message Type          | Invite.Response.3xx              |
| Condition             | Header.Diversion.URL.Type == '2' |
| Action Subject        | header.diversion.url.type        |
| Action Type           | Modify                           |
| Action Value          | '1'                              |

Signaling & Media->Message Manipulation->Message Manipulations

| Manipulation Set ID 0 |       |
|-----------------------|-------|
| Parameter             | Value |

|                |                                                                      |
|----------------|----------------------------------------------------------------------|
| Message Type   | Invite.Response.3xx                                                  |
| Condition      | Header.Diversion.URL.Type == '2'                                     |
| Action Subject | Header.Diversion.URL.Host.Name                                       |
| Action Type    | Add                                                                  |
| Action Value   | '10.10.40.30' (pls replace with the IP address of the LAN interface) |

Signaling & Media->Core Entities->IP Groups (in addition to the necessary, but not described parameters)

| Name ESIP                     |       |
|-------------------------------|-------|
| Parameter                     | Value |
| Outbound Message Manipulation | 0     |

## 4.2 Manipulation & Routing

Signaling & Media->Gateway->Manipulation->Dest Number Tel->IP

| Rule E164_INTERNAT        |       |
|---------------------------|-------|
| Parameter                 | Value |
| Destination Prefix        | 00*   |
| Destination IP Group      | ESIP  |
| Stripped Digits From Left | 2     |
| Prefix to Add             | +     |

| Rule E164_NATIONAL        |                |
|---------------------------|----------------|
| Parameter                 | Value          |
| Destination Prefix        | 0[1-9]xxxxxxxx |
| Destination IP Group      | ESIP           |
| Stripped Digits From Left | 1              |
| Prefix to Add             | +41            |

| Rule E164_COMBOX          |              |
|---------------------------|--------------|
| Parameter                 | Value        |
| Destination Prefix        | 0860xxxxxxxx |
| Destination IP Group      | ESIP         |
| Stripped Digits From Left | 1            |
| Prefix to Add             | +41          |

| Rule E164_EMERGENCY       |                     |
|---------------------------|---------------------|
| Parameter                 | Value               |
| Destination Prefix        | 11x, 144, 145, 141x |
| Destination IP Group      | ESIP                |
| Stripped Digits From Left | 0                   |
| Prefix to Add             | +41                 |

| Rule E164_SERVICE |  |
|-------------------|--|
|-------------------|--|

| Parameter                 | Value |
|---------------------------|-------|
| Destination Prefix        | 1x*   |
| Destination IP Group      | ESIP  |
| Stripped Digits From Left | 0     |
| Prefix to Add             | +41   |

Optional:

| Rule E164_GEOLOC          |        |
|---------------------------|--------|
| Parameter                 | Value  |
| Destination Prefix        | 086756 |
| Destination IP Group      | ESIP   |
| Stripped Digits From Left | 1      |
| Prefix to Add             | +41    |

Signaling & Media->Gateway->Source Number IP->Tel

| Rule EXPL_to_IMPL_+41     |       |
|---------------------------|-------|
| Parameter                 | Value |
| Source Phone Pattern      | +41*  |
| Source IP Group           | ESIP  |
| Stripped Digits From Left | 3     |
| Prefix to Add             | 0     |

| Rule EXPL_to_IMPL         |       |
|---------------------------|-------|
| Parameter                 | Value |
| Source Phone Pattern      | +*    |
| Source IP Group           | ESIP  |
| Stripped Digits From Left | 1     |
| Prefix to Add             | 00    |

Signaling & Media->Gateway->Routing->Tel->IP Routing

| Rule FXS_to_ESIP     |       |
|----------------------|-------|
| Parameter            | Value |
| Destination IP Group | ESIP  |

Signaling & Media->Gateway->Routing->IP->Tel Routing

| Rule ESIP_to_FXS          |                  |
|---------------------------|------------------|
| Parameter                 | Value            |
| Source SIP Interface      | any              |
| Destination Phone Pattern | +4131924147[0-4] |
| Trunk Group ID            | 1                |
| Source IP Group           | ESIP             |
| Trunk ID                  | 1                |

Signaling & Media->Gateway->Trunk Group Settings

| Name by_number         |                      |
|------------------------|----------------------|
| Parameter              | Value                |
| Trunk Group ID         | 1                    |
| Channel Select Mode    | By Dest Phone Number |
| Used By Routing Server | Yes                  |
| Serving IP Groupe      | ESIP                 |

#### 4.3 Coder & Profiles

Signaling & Media->Coders and Profiles->Tel Profile

| Parameter                | Value              |
|--------------------------|--------------------|
| Profile ID               | 1                  |
| Profile Name             | G.711 only         |
| Fax Signalling Method    | No Fax             |
| Progress Indicator to PI | No PI              |
| Echo Canceler            | Disable            |
| Coder Group              | AudioCoderGroups_0 |

Optional: Tel Profile for T.38: not necessary as default configured behavior is T.38 relay

| Parameter             | Value      |
|-----------------------|------------|
| Profile ID            | 2          |
| Profile Name          | T.38       |
| Fax Signalling Method | T.38 Relay |

#### 4.4 Fax part

Signaling & Media->Media->Fax/Mode/CID Setting

| Parameter                 | Value            |
|---------------------------|------------------|
| Fax Transport Mode        | Disable          |
| V.22 Modem Transport Type | Disable          |
| V.23 Modem Transport Type | Disable          |
| V.32 Modem Transport Type | Disable          |
| V.34 Modem Transport Type | Disable          |
| Fax Relay Max Rate (bps)  | 9600bps          |
| Fax Relay ECM Enable      | Disable          |
| Fax CNG Mode              | Send on CNG tone |

Signaling & Media->Gateway->Gateway General Settings

| Parameter             | Value        |
|-----------------------|--------------|
| Fax Signalling Method | Fax Fallback |

|                     |                      |
|---------------------|----------------------|
| SIP T.38 Version    | Version 0            |
| Channel Select Mode | By Dest Phone Number |

Optional: To test T.38 with sending fax sending T.38 reINVITE immediately after establishment of the call:  
Signaling & Media->Media->Fax/Mode/CID Setting

| Parameter         | Value                    |
|-------------------|--------------------------|
| CNG Detector Mode | Relay instead of Disable |

#### 4.5 Voice part

Signaling & Media->Media->Fax/Mode/CID Setting

| Parameter                | Value |
|--------------------------|-------|
| Caller ID Transport Type | Relay |

Signaling & Media->Gateway->DTMF & Supplementary->Supplementary Services Settings

| Parameter        | Value  |
|------------------|--------|
| Enable Caller ID | Enable |

#### 4.6 Trunk & Number assignment (example)

Signaling & Media->Gateway->Trunk Groups

(In our configuration, +41319241470 & 71 are reserved for DECT devices, 72 for G.711 only fax and 73 for T.38 fax)

| GROUP INDEX | MODULE       | FROM TRUNK | TO TRUNK | CHANNELS | PHONE NUMBER | TRUNK GROUP ID | TEL PROFILE NAME |
|-------------|--------------|------------|----------|----------|--------------|----------------|------------------|
| 1           | Module 2 FXS |            |          | 1        | +41319241470 | 1              | None             |
| 2           | Module 2 FXS |            |          | 2        | +41319241471 | 1              | None             |
| 3           | Module 2 FXS |            |          | 3        | +41319241472 | 1              | G.711 only       |
| 4           | Module 2 FXS |            |          | 4        | +41319241473 | 1              | None             |

#### 4.7 Optional customer defined settings

The below settings are not mandatory and can be freely defined by the customer. The below examples are taken from the homologation's testing environment.

##### 4.7.1 Keypad Features

Signaling & Media->Gateway->Analog Gateway->Keypad Features

| Parameter                     | Value |
|-------------------------------|-------|
| Hook-Flash Code               | *56   |
| Forward Unconditional         | *21   |
| Forward No Answer             | *61   |
| Forward On Busy               | *67   |
| Forward Deactivate            | #20   |
| Restricted Caller ID Activate | *98   |

Restricted Caller ID Deactivate

#98

#### 4.7.2 Caller Display Information

- Name of the FXS subscriber, i.e., the name which will be sent in the FROM field beside of the number in case of outgoing call.

| INDEX | MODULE | PORT | PORT TYPE | DISPLAY STRING          | PRESENTATION |
|-------|--------|------|-----------|-------------------------|--------------|
| 0     | 2      | 1    | FXS       | DECT VSG                | Allowed      |
| 1     | 2      | 2    | FXS       | DECT VSG                | Allowed      |
| 2     | 2      | 3    | FXS       | Fax G.711 only          | Allowed      |
| 3     | 2      | 4    | FXS       | Fax both G.711 and T.38 | Allowed      |

- In case of outgoing call and the presentation of the Caller ID must be restricted, the configuration can be achieved either over the web GUI or with the keypad feature:

| GENERAL   |     | CALLER DISPLAY |            |
|-----------|-----|----------------|------------|
| Index     | 0   | Display String | DECT VSG   |
| Module    | 2   | Presentation   | Restricted |
| Port      | 1   |                |            |
| Port Type | FXS |                |            |

Procedure on the FXS subscriber to activate the Restricted Caller ID:

\*98 <hear the confirmation tone>

Procedure on the FXS subscriber to deactivate the Restricted Caller ID:

#98 <hear the confirmation tone>

#### 4.7.3 Caller ID Permissions

In case of incoming call, the caller ID is per default displayed on the FXS subscriber. This setting is enabled for all the ports, but it is possible to disable this feature per FXS subscriber:

Caller ID Permissions (4)

[Edit](#)

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| INDEX | MODULE | PORT | PORT TYPE | CALLER ID               |
|-------|--------|------|-----------|-------------------------|
| 0     | 2      | 1    | FXS       | <a href="#">Disable</a> |
| 1     | 2      | 2    | FXS       |                         |
| 2     | 2      | 3    | FXS       |                         |
| 3     | 2      | 4    | FXS       |                         |

#0

[Edit](#)

| GENERAL   |     |
|-----------|-----|
| Module    | 2   |
| Port      | 1   |
| Port Type | FXS |

| CALLER ID |                         |
|-----------|-------------------------|
| Caller ID | <a href="#">Disable</a> |

## 4.7.4 Call Waiting

Signaling & Media->Gateway->DTMF & Supplementary->Supplementary Services Settings

| Parameter           | Value   |
|---------------------|---------|
| Enable Call Waiting | Disable |

Call Waiting (4)

[Edit](#)

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| INDEX | MODULE | PORT | PORT TYPE | ENABLE CALL WAITING     |
|-------|--------|------|-----------|-------------------------|
| 0     | 2      | 1    | FXS       | <a href="#">Enable</a>  |
| 1     | 2      | 2    | FXS       | <a href="#">Disable</a> |
| 2     | 2      | 3    | FXS       | <a href="#">Disable</a> |
| 3     | 2      | 4    | FXS       | <a href="#">Disable</a> |

## 4.7.5 Call Forward

Call Forward (4)

[Edit](#)

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| INDEX | MODULE | PORT | PORT TYPE | TYPE          | FORWARD DESTINATION | NO REPLY TIME |
|-------|--------|------|-----------|---------------|---------------------|---------------|
| 0     | 2      | 1    | FXS       | Unconditional | 0319241421          | 30            |
| 1     | 2      | 2    | FXS       | Deactivate    |                     | 30            |
| 2     | 2      | 3    | FXS       | Deactivate    |                     | 30            |
| 3     | 2      | 4    | FXS       | Deactivate    |                     | 30            |

Procedure on the FXS subscriber to activate an unconditional forward to 0319241421:

\*21 <hear the dial tone> 0319241421 <terminate with #> <hear the confirmation tone>

Procedure on the FXS subscriber to activate a forward on no answer to 0319241421:

\*61 <hear the dial tone> 0319241421 <terminate with #> <hear the confirmation tone>

Procedure on the FXS subscriber to activate a forward on busy to 0319241421:

\*67 <hear the dial tone> 0319241421 <terminate with #> <hear the confirmation tone>

Procedure on the FXS subscriber to deactivate any kind of forward:

#20 <hear the confirmation tone>

For the forward on no answer, the timer must be adjusted, if needed, over the web GUI:

| GENERAL   |             | CALL FORWARD        |              |
|-----------|-------------|---------------------|--------------|
| Index     | 0           | Forward Destination | * 0319241421 |
| Module    | 2           | No Reply Time       | 10           |
| Port      | 1           |                     |              |
| Port Type | FXS         |                     |              |
| Type      | * No answer |                     |              |

#### 4.7.6 Alternative number for outgoing calls

It is possible to use another number than the regular number for outgoing call. To achieve that, a source number manipulation must be created.

Example to use the alternative number +41800800800 instead of the regular number +41319241470:

| GENERAL                   |                | ACTION                     |                |
|---------------------------|----------------|----------------------------|----------------|
| Index                     | 0              | Stripped Digits From Left  | 0              |
| Name                      |                | Stripped Digits From Right | 0              |
| MATCH                     |                | Number of Digits to Leave  | * 0            |
| Source Trunk Group        | -1             | Prefix to Add              | * +41800800800 |
| Source Phone Pattern      | * +41319241470 | Suffix to Add              |                |
| Destination Phone Pattern | *              | TON                        |                |
|                           |                | NPI                        |                |
|                           |                | Presentation               |                |

Cancel APPLY



## 5 SIP Trunk requirements and recommendations

The Swisscom requirements described in this chapter are a referencing the SIP Trunk UNI Guide for Enterprise SIP [1].

### 5.1 URI & UIFN Addressing

A number is a string of decimal digits that uniquely indicates the network termination point (hereafter called dial string). The dial string contains the information necessary to route the call to this point. Over-decadic digits are NOT supported. A dial string MUST therefore include only the following characters: 0-9,\*,#,+ ('#' character is represented as "%23", based on RFC 2396). Dial string is expressed in the User-URI Portion of the SIP URI according to RFC 3966. The number will be transported in the user part of the URI.

A name is a string of characters acting as a more readable alias for a given number. It MAY or MAY NOT be presented to the terminating user depending on the target system. Characters used for Names Strings SHALL use the ISO 10646 character set in UTF-8 encoding (RFC 2279). The name will be transported in the display part of the URI.

#### 5.1.1 SIP URI Addressing scheme

Numbering and Addressing Scheme used in all SIP Headers MUST be E.164-based and follow the format defined in RFC3966 Chapter 5.1.4 Global Number format. These numbers MUST be expressed using SIP URI. TEL URI is NOT supported in any header.

In order to successfully setup a call, all telephone number used in the SIP URI MUST be a valid E.164 number preceded with the "+" character and the **user** parameter value "**phone**" MAY be present as described in RFC 3261 section 19.1.1.

In Summary:

- Standard Swiss Geo- and None-Geo Numbers MUST be prefixed by **+41** (e.g., +41582219911)
- Domestic Toll-Free, Shared Cost and Premium MUST be prefixed by **+41** (e.g., +41800800800)
- International Numbers MUST be prefixed by the corresponding Country Code **[+CC]** (e.g., +33142764040)
- International Toll-Free MUST be prefixed by the corresponding Country Code **[+CC]** (e.g., +44800800800)
- Universal International Free phone Number (UIFN) MUST be prefixed by **+** (e.g., +80012345678)
- All 3- and 4 digits Short-Numbers MUST be prefixed by **+41** (e.g., +41117, +411414)

All other formats are NOT supported (i.e., national, subscriber, private, etc.)

The above Requirements SHALL be applied to ALL SIP Headers containing URI Information but MUST be applied at least to the R-URI, FROM, TO, PAI, History-Info and Diversion Headers. If not properly applied on Optional Headers such as P-Preferred-Identity, proper CLID Propagation and/or per-subscriber Billing MAY not be possible. This requirement is valid for SIP requests and responses (i.e., 200OK or 181CallsBeingForwarded).

When sending a INVITE Request, the UA SHALL ensure that Request-URI set to the same URI value of the To header. Examples of a **valid** SIP URI are:

<sip:+41582219911@xxx.xxx.xxx.xxx> <sup>1</sup>

<sip:+41582219911@xxx.xxx.xxx.xxx;user=phone> <sup>1</sup>

<sip:+41582219911@domain.com> <sup>2</sup>

<sip:+41582219911@domain.com;user=phone> <sup>2</sup>

<sip:anonymous@anonymous.invalid> <sup>3</sup>

Examples of **invalid** SIP URI are:

<sip:0582219911@domain.com;user=phone>  
<sip:0041582219911@domain.com;user=phone>  
<sip:41582219911@domain.com;user=phone>  
<sip:@domain.com>

<sup>1</sup> xxx.xxx.xxx.xxx = represents a valid IPv4 Address and SHALL be the public IP-Address of the Originating PBX/SBC in case of FROM/PAI/DIV/Hi and sent-to IP-Address in the R-URI and TO Header.

<sup>2</sup> *domain.com* represents either the domain name that matches the corresponding SIP domain of the originating or the terminating Network (depending on Header) and is subject of bilateral peering Partners. In PAI Header, *domain.com* SHALL reflect the origin Domain of the calling-party if applicable.

<sup>3</sup> applies to the FROM Header ONLY in case of CLIR.

## 5.2 Out of the session keepalive: SIP OPTIONS or INFO

SIP OPTIONS, respectively INFO, MUST be supported and accepted by the UA. The UA MUST respond with a *200OK* or *483TooManyHops* in case the Hop Count has been set to 0 as done by the Swisscom Border Controller.

For sending SIP OPTIONS or INFO, if possible and/or supported by the UA, the Max-Forwards Header SHALL be set to 0 to terminate SIP Probing right at the ingress point. For all other SIP Messages, the Max-Forwards Header SHALL be left to the default value (70) as defined in Section 11 of RFC 3261.

Since the SIP OPTIONS or INFO are intended as a probing mechanism for SIP reachability between Swisscom border element and UA it is recommended that the URI of the request is ignored (as example UA should not attempt to identify a local user matching the R-URI). SIP OPTIONS SHALL be addressed to the sent-to IP-Address.

The UA MAY provide in the SIP response to OPTIONS or INFO request his capabilities (Allow and Supported header as well SDP including all possible media capabilities). Currently these capabilities are not actively used in call processing but may be part of future extensions.

**Swisscom recommendation:**

To speed up failover in redundant environments (with Multi SAP, SDT-1 or Multi Server Enterprise SIP service options), it is recommended to set the SIP OPTIONS timers to the following values:

- Up-Interval 10sec
- Down-Interval 10sec
- Retry count 2

## **5.3 DTMF**

### **5.3.1 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 be guaranteed by either side.

In case of DTMF transmission the SDP MUST contain the *rtptime* 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.

### **5.3.2 Receiving (from Swisscom)**

To insure (backward) compatibility with a system that doesn't 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:**

If 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.