How To Guide

SIP Trunking Configuration
Using the SIP Trunk Page

For the Ingate SIParators® and Firewalls using software release 4.10.1 or later

Revision 1.4

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1   Using the SIP Trunk Configuration Page in the Ingate

The SIP Trunk page is new from version 4.9.0 of the Ingate firmware.

Using the SIP Trunk page will always invoke the B2BUA for the connection of the PBX to the service provider's SIP Trunk and offers the following advantages:
- The IP-PBX only talks to the Ingate
- Wider separation between the PBX and the Trunk Service
- The SIP Trunking Service Credentials (User IDs and Passwords) are stored only in the Ingate
- More SIP normalization possibilities
- Individual or main CallerID (number) presented to the called party
- If a new SIP Trunking Service platform is introduced, only the Ingate has to be reconfigured.

There can be up to nine SIP Trunk pages with individual PBXs, trunks and interoperability settings.

When using the SIP Trunk page, there will be a clear demarcation point for the SIP Trunking service hand off.

Using the SIP Trunk page improves the previous way of configuring the Ingate. It offers simplicity, additional functionality and a wider support of various PBXs and Trunk Services. (It is still possible to configure SIP Trunking the previous way, just leaving this SIP Trunking page empty.)

1.1   Both the PBX with its Phones and Other SIP Clients Can Use the SIP Trunk

The SIP Trunk page also offers a simple way to allow SIP clients (SIP phones and Soft SIP clients for PCs) to register to the Ingate and use the SIP Trunk. This is done in addition to supporting the PBX and its extension phones.

1.2   Licenses/Modules Required for the SIP Trunk Page

You can have up to nine SIP Trunk pages for different PBXs and or SIP Trunk Services, but usually only one is required. The SIP Trunking Module license is required to use SIP Trunk pages and one page is included with that license. To get more SIP Trunk pages, you buy Additional Trunk Group licenses.

1.3   Steps to Configure SIP Trunking

There are four steps to connect the PBX to the SIP Trunking service. Each of these steps is simple and straightforward:
- Define on the Dial Plan page which outgoing calls to send to the SIP Trunk
- Enter the SIP Service parameters (at the top of the SIP Trunk page)
- Enter the PBX parameters (at the bottom of the SIP Trunk page)
- Define the registration and authentication towards the Trunk Service and the number routing between the Trunk Service and the PBX (in the middle of the SIP Trunk page)

Note that there are help texts available in the Ingate for each setting – just press the "Help" links on the SIP Trunking page.

The Ingate Startup Tool TG, will use the SIP Trunk page when configuring SIP Trunking. (Use the previous Ingate Startup Tool - without the "TG" postfix - if you want to configure the previous way, without using the SIP Trunk page.)

2   Setting up SIP Trunking

This section describes the functioning and set-up of the SIP Trunk page in the Ingate.
Please remember that these settings, as well as most of the other ones in this description, usually are done by the Startup Tool TG and that this description is provided for reference.

2.1   Defining Outgoing Call Handling in the Dial Plan

Outgoing calls are processed through the Dial Plan, which must be **On**. At a minimum, the fields exemplified below must be entered. The actual Dial Plan table is searched line by line from the top for a match from the PBX of the dialed number where after it is forwarded to the SIP Trunk page.

1) Calls from the PBX connected to network "ShoreTel"
2) …with a any dialed number send sent to this unit (regular expression (.*)@10.100.0.13)
3) …will be forward for further processing to "Gamma Trunk"
4) …which is defined on SIP Trunk page 1

2.2 The SIP Trunk Page

The SIP Trunk pages are found under SIP Trunks. Several SIP Trunk pages may be defined if you have several PBXs or Trunk Services. You need to purchase Additional Trunk Group licensees to get more than one SIP Trunk page. Details are found below.

2.2.1 Entering the SIP Trunking Service Parameters

For connecting to the exemplified TService, only three of the many parameters have to be filled-in.

1) Service Provider domain, here "tservice.com", which is the host part of the SIP address of the account such as 1306770800@tservice.com. (Enter two comma separated domains or IP addresses for failover configuration.)

2) Outbound Proxy is where the SIP requests for this trunk service are to be sent. Here a fixed IP address is used (but a domain name is preferred, especially if the service provider has a DNS SRV record).

3) This service provider requires a P-Preferred-Identity header to be used to show the caller's number as caller ID. The actual header content is taken from the Identity field in the tables below.

For details, please see the (Help) texts in the Ingate GUI and last in this document.

2.2.2 Entering the PBX Parameters

Here you enter the basics relating to the IP PBX to be connected to the SIP Trunking Service. Other settings may be required, e.g. on the SIP Interoperability page and other pages.

1) The Ingate needs to know at which address the PBX is located, which can be entered as an IP address, a domain name or a SIP registration by the PBX. Here you enter the IP address if the PBX is at a fixed IP address. (Enter two comma separated domains or IP addresses in the "PBX Domain Name" field for failover configuration.)

2) Here you can set up a SIP account, at which the PBX registers itself to show where it can be contacted. If the PBX is located on a fixed IP address, it is recommended that you *also* enter that one (as done under *1)* above).

By entering User ID and Password, the PBX will be forced to Authenticate itself when registering to the Ingate. The User ID and Password are also required if the Ingate is configured to authenticate users placing outgoing calls (which is done in the Dial Plan by selecting "Auth & Forward" instead of just "Forward" under **Action**).

3) You also need to specify to which defined network the PBX is connected to.

Setup for the PBX (Help)

○ Use PBX from other SIP trunk
◉ Define PBX settings

PBX Name: thePBX (Descriptive name)

Use alias IP address: - ▼ (Forces this source address from our side)

PBX Registration SIP Address	Authentication		PBX IP Address		PBX Domain Name
	User ID	Password	DNS Name or IP Address	IP Address	
123456@10.100.0.13	thepbx	Change Password	10.100.0.40	10.100.0.40	

2) .... *1)*

(At least one of PBX Registration, IP address or Domain Name is required to locate the PBX)

3) PBX Network: ShoreTel ▼

Signaling transport: - ▼ ('-' = Automatic)

Port number: [ ]

Match From Number/User in field: From URI ▼

To header field:
◉ Same as Request-URI
○ Copy from Trunk
○ Initial Request-URI
○ as entered:

Remote Trunk Group Parameters usage: - ▼ ('-' = Don't use TGP)

Local Trunk Group Parameters usage: - ▼ ('-' = Don't use TGP)

Other PBXs (than here exemplified) may require more of the available settings. For details, please see the (Help) texts in the Ingate GUI and last in this document.

2.2.3 Registration to and Authentication for the Trunking Service (General Overview)

In the section of the SIP Trunk page shown below, you configure whether the Ingate shall register to some of the Trunk Service accounts (**A:**) (e.g. to one or to all numbers) and enter the Used ID and Password for the Trunk Service is (**B:**). The SIP Trunking service provider shall have given you these details, together with the telephone numbers (DID numbers) to use.

2.2.3.1 No Registration to the SIP Trunking Service

This is typical for services that accepts usage of service from the user's specific IP address and don't require any authentication password. All rows in the **Reg.** column (**A:**) shall be "No" and if you have not received any authentication Password from your service provider, leave the **Authentication** column empty.

2.2.3.2 One Registration to your Main Account from the Service Provider

In this case you set Reg. to Yes (**A:**) and enter the authentication **User ID** and **Password** (**B:**) that you have received from your service provider on the Main Trunk Line row. The authentication **User ID** is often the same as the telephone number.

2.2.3.3 Registrations for Each DID Numbers from the Service Provider

In this case, there need to be a row for each DID number in the tables below and all those rows have to have Yes in the **Reg.** column (**A:**). If the service provider has given you a single authentication **User ID** and **Password** (**B:**) pair, enter that on the Main Trunk Line and it will be used for all numbers. In case you have been given individual User IDs and Passwords, enter them on the corresponding rows.

	A:	**C:** Calls from a specific user will use the trunk line defined here.	**B:** Credentials for Trunk Service	**D:** Incoming calls will be sent to a specific user here.

Main Trunk Line (Help)

No.	Reg	Outgoing Calls				Authentication		Incoming Calls		
			Display Name	User Name	Identity	User ID	Password	Incoming Trunk Match	Forward to	
1	Yes ▼	LEAVE EMPTY!		1305670700	1305670700@tservice.c	305670700	Change Password			

PBX Lines (Help)

No.	Reg	Outgoing Calls				Authentication		Incoming Calls		Delete Row
		From PBX Number/User	Display Name	User Name	Identity	User ID	Password	Incoming Trunk Match	Forward to PBX Account	
1	No ▼	13056707([0-8][0-9])	13056707$1	13056707$1			Change Password	0(13056707[0-8][0-9])	$1	☐
2	No ▼	anonymous		anonymous@anonymou			Change Password			☐

Add new rows 1 rows.

SIP Lines (Help)

No.	Reg	Outgoing Calls				Authentication		Incoming Calls		Delete Row
		From SIP Number/User	Display Name	User Name	Identity	User ID	Password	Incoming Trunk Match	Forward to SIP Account	
1	No ▼	steven	Steven Brown	01305670790			Change Password	01305670790	steven	☐
2	No ▼	/+1305670799	Fax	01305670799			Change Password	01305670799	+1305670799	☐
3	No ▼	pda(9[0-9])		13056707$1			Change Password	013056707(9[0-9])	pda$1	☐

Add new rows 1 rows.

2.2.4 Defining Outgoing and Incoming Number Routing (General Overview)

Column Descriptions:

In the section of the SIP Trunk page shown above, you also configure how outgoing calls from the PBX are routed to the Trunk Service (***C:***) and how incoming calls from the trunk service are routed to the PBX (***D:***).

The Identity column is used when the trunking service requires P-Asserted-Identity or P-Preferred-Identity to be used. See also the "Service Provider domain is trusted (for P-Asserted-Identity)" and "Use P-Preferred-Identity (instead of P-Asserted-Identity)" among the SIP Trunking Service settings in 2.2.1.

For an <u>outgoing call</u>, the caller's number (the user part of the From header) will - row by row, from top to bottom - be checked against the numbers or regular expressions entered in the column **From PBX/SIP Number/User**. A match will cause the trunk account under **User Name** to be used for the outgoing call (***C:***).

An <u>incoming call</u> will be checked against the numbers or regular expressions entered in the column **Incoming Trunk Match**. A match will cause the call to be forwarded to the PBX or SIP user specified in the "**Forward to…**"-accounts.

Regular Expressions:

Notice that you can use numbers, sip user names or regular expressions in the column **From PBX/SIP Number/User** and in the column **Incoming Trunk Match**. More information is found in the How To Guide: How To use Regular Expressions.

The results of subexpressions - (…) in the regular expression - can be used as $0, $1 etc. in the following column(s):
- The results of subexpressions from a match in the column **From PBX/SIP Number/User** can be used in the columns **Display Name, User Name** and **Identity**. See (***C:***).
- The results of subexpressions from a match in the column **Incoming Trunk Match** can be used in the column **Forward to…**. See (***D:***).

Since the regular expression syntax is used, some characters has to be preceded by the "\" escape character. E.g. phone number +14081234567 has to be entered as "\+14081234567".

Generic Header Manipulation:

If needed, you can use Generic Header Manipulation in the columns: **User Name** (for outgoing calls) and in column **Forward to…** (for incoming calls). More information is found in the How To Guide: How To use Generic Header Manipulation

Row Descriptions:

The Main Trunk Line settings (**E:**) are used if the caller (**From PBX/SIP Number/User**) or credentials (**User ID** and **Password**) are not defined further down. For services that specify one main number or account, this is where those settings are entered. At (**F:**) you enter the routing between the external (DID) numbers on the SIP Trunk and the numbers used by the PBX. At (**G:**) you can link SIP Clients (e.g. SIP Phones and Soft PC SIP Clients) registered to the Ingate itself or else (not belonging to the PBX), to trunk numbers.

E: Main Trunk Line   (Help)

No.	Reg	Outgoing Calls				Authentication		Incoming Calls	
			Display Name	User Name	Identity	User ID	Password	Incoming Trunk Match	Forward to
1	Yes ▼	LEAVE EMPTY!		1305670700	1305670700@tservice.c	305670700	Change Password		

F: PBX Lines   (Help)

No.	Reg	Outgoing Calls				Authentication		Incoming Calls		Delete Row
		From PBX Number/User	Display Name	User Name	Identity	User ID	Password	Incoming Trunk Match	Forward to PBX Account	
1	No ▼	13056707([0-8][0-9])	13056707$1	13056707$1			Change Password	0(13056707[0-8][0-9])	$1	☐
2	No ▼	anonymous		anonymous@anonymou			Change Password			☐

Add new rows 1 rows.

G: SIP Lines   (Help)

No.	Reg	Outgoing Calls				Authentication		Incoming Calls		Delete Row
		From SIP Number/User	Display Name	User Name	Identity	User ID	Password	Incoming Trunk Match	Forward to SIP Account	
1	No ▼	steven	Steven Brown	01305670790			Change Password	01305670790	steven	☐
2	No ▼	\+1305670799	Fax	01305670799			Change Password	01305670799	+1305670799	☐
3	No ▼	pda(9[0-9])		13056707$1			Change Password	013056707(9[0-9])	pda$1	☐

Add new rows 1 rows.

For details, please see the (Help) texts in the Ingate GUI and last in this document.

2.2.5 Typical Settings when the PBX is Configured to use the External (DID) Numbers

1) In this example, there should be no registration to the trunking service. This is typical for a service that simply accepts usage of the trunk from the specific IP address of the user.

2) Select one of the numbers of the trunking service (of the DID numbers) as your main number and enter it on this line. For this trunking service, there is no password since it authenticates the user by its IP address. Therefore, simply leave **User ID** and **Password** empty. Calls from PBX users that are NOT entered further down in the **From PBX/SIP Number/User** column, will use this trunk line and the main number will be shown to the called party (as Caller ID).

3) Enter the other DID numbers you want to use with the PBX on these line. The number under **User Name** will be shown to the called party as Caller ID. If the trunking service (in order to place an anonymous call) requires anonymous@anonymous.invalid (or something else) in the From header, the user name part of the From header from the PBX (e.g. anonymous or Private) has to be entered on a separate row as shown.

4) Incoming calls will be forwarded to the PBX number (user) defined in the **Forward to…** column. Here all DID numbers 01305670700 to 01305670789 are mapped to the PBX numbers 1305670700 to 1305670789 on a single line.

The mapping between the external (DID) numbers and the PBX numbers does not have to be as regular as shown here. One can add prefixes and map to any number, or even use the extension numbers of the PBX, see next section 2.2.6.

Under SIP Lines, examples of are shown how SIP Clients not belonging to the PBX, still can be connected to the PSTN and use DID numbers from the SIP trunking service. See section 2.2.7.

Main Trunk Line (Help)

No.	Reg		**Outgoing Calls**			**Authentication**		**Incoming Calls**	
			Display Name	User Name	Identity	User ID	Password	Incoming Trunk Match	Forward to
1	No	LEAVE EMPTY!		1305670700			Change Password		

PBX Lines (Help)

No.	Reg	**Outgoing Calls** From PBX Number/User	Display Name	User Name	Identity	**Authentication** User ID	Password	**Incoming Calls** Incoming Trunk Match	Forward to PBX Account	Delete Row
1	No	13056707([0-8][0-9])	13056707$1	13056707$1			Change Password	0(13056707[0-8][0-9])	$1	☐
2	No	anonymous		anonymous@anonymou			Change Password			☐

Add new rows 1 rows.

SIP Lines (Help)

No.	Reg	**Outgoing Calls** From SIP Number/User	Display Name	User Name	Identity	**Authentication** User ID	Password	**Incoming Calls** Incoming Trunk Match	Forward to SIP Account	Delete Row
1	No	steven	Steven Brown	01305670790			Change Password	01305670790	steven	☐
2	No	\+1305670799	Fax	01305670799			Change Password	01305670799	+1305670799	☐
3	No	pda(9[0-9])		13056707$1			Change Password	013056707(9[0-9])	pda$1	☐

Add new rows 1 rows.

In the PBX:
With the settings above in the Ingate, you need to enter all phone numbers (DID numbers) used with the trunk in the PBX and also map each of those numbers to the extension numbers the PBX uses.

2.2.6 Settings when the PBX Only Uses Internal Extension Numbers

The settings are similar to those in the previous section 2.2.5, but the PBX users are referenced by their extension numbers 00 to 89 (instead of the DID numbers 1305670700 to 1305670789) as exemplified at *A:* and *B:*.

Main Trunk Line (Help)

No.	Reg	Outgoing Calls				Authentication		Incoming Calls	
			Display Name	User Name	Identity	User ID	Password	Incoming Trunk Match	Forward to
1	Yes ▾	LEAVE EMPTY!		1305670700	1305670700@tservice.c	305670700	Change Password		

PBX Lines (Help)   *A:*   *B:*

No.	Reg	From PBX Number/User	Outgoing Calls			Authentication		Incoming Calls		Delete Row
			Display Name	User Name	Identity	User ID	Password	Incoming Trunk Match	Forward to PBX Account	
1	No ▾	088	Operator	1305670788			Change Password			☐
3	No ▾	[0-8][0-9]	$0	13056707$1			Change Password	013056707([0-8][0-9])	$1	☐
4	No ▾	anonymous		anonymous@anonymou			Change Password			☐

Add new rows 1 rows.

SIP Lines (Help)

No.	Reg	From SIP Number/User	Outgoing Calls			Authentication		Incoming Calls		Delete Row
			Display Name	User Name	Identity	User ID	Password	Incoming Trunk Match	Forward to SIP Account	
1	No ▾	steven	Steven Brown	01305670790			Change Password	01305670790	steven	☐
2	No ▾	\+1305670799	Fax	01305670799			Change Password	01305670799	+1305670799	☐
3	No ▾	pda(9[0-9])		13056707$1			Change Password	013056707(9[0-9])	pda$1	☐

Add new rows 1 rows.

In the PBX:

With the settings suggested in this section, the Ingate will forward calls directly to the extension numbers used by the PBX. Thus, no DID numbers (allocated by the service provider) have to be entered in the PBX and the PBX is expected to show the extension number of the caller in its From header.

2.2.7 Using SIP Clients Registered to the Ingate with the SIP Trunk

In this example, we allow other SIP clients (SIP Phones or soft PC Clients) than the PBX extensions, to connect to the PSTN and use DID numbers from the SIP Trunk.

1) Steven Brown has a soft SIP PC client with SIP address steven@<Ingate LAN IP address> and is mapped to the DID number 01305670790, that he can use for both outgoing and incoming calls.

2) An IP fax machine has been connected to the LAN and registered to the Ingate. The fax machine is set up with sip address +1305670799@<Ingate LAN IP address>. Notice the escape character "\" in the **From SIP Number/User** column, since the "+" character has a special meaning in regular expressions.

3) Eight PDA with SIP names pda91@<Ingate LAN IP address> to pda98@<Ingate LAN IP address> are registered to the Ingate and will be mapped to DID numbers 01305670791 to 01305670791 for both outgoing and incoming calls.

Main Trunk Line (Help)

No.	Reg		Outgoing Calls			Authentication		Incoming Calls	
			Display Name	User Name	Identity	User ID	Password	Incoming Trunk Match	Forward to
1	Yes ▾	LEAVE EMPTY!		1305670700	1305670700@tservice.c	305670700	Change Password		

PBX Lines (Help)

No.	Reg	From PBX Number/User	Display Name	User Name	Identity	User ID	Password	Incoming Trunk Match	Forward to PBX Account	Delete Row
1	No ▾	088	Operator	1305670788			Change Password			☐
3	No ▾	[0-8][0-9]	$0	1305670$1			Change Password	013056707([0-8][0-9])	$1	☐
4	No ▾	anonymous		anonymous@anonymou			Change Password			☐

Add new rows 1 rows.

SIP Lines (Help)

	No.	Reg	From SIP Number/User	Display Name	User Name	Identity	User ID	Password	Incoming Trunk Match	Forward to SIP Account	Delete Row
1)	1	No ▾	steven	Steven Brown	01305670790			Change Password	01305670790	steven	☐
2)	2	No ▾	\+1305670799	Fax	01305670799			Change Password	01305670799	+1305670799	☐
3)	3	No ▾	pda(9[0-9])		1305670$1			Change Password	013056707(9[0-9])	pda$1	☐

Add new rows 1 rows.

These SIP clients each require a Register **Account Type** to be set-up on the **SIP Traffic > Local Registrar** page (for older versions than 4.10 software this is on the **SIP Traffic > SIP Accounts** page).

You can also allow other SIP clients (not belonging to the PBX and not registered to the Ingate) to use the trunk. Then simply enter the client's full SIP address, e.g. gustav@smartcompany.com or 08123456@sipworld.org, in the **Forward to SIP Account** column.

To allow SIP clients, like Steven's Soft PC client and the other above, to use the Ingate as their SIP server, the SIP Registrar Module License is required.

3 Field Descriptions for the SIP Trunking Page

These descriptions are also found in the Ingate product GUI help texts.

3.1 SIP Trunking Service

This section contains settings for the interface towards the ITSP, the SIP Trunk.

Use parameters from other SIP trunk / Define SIP trunk parameters - Determines whether the settings for the SIP Trunking Service should be taken from this page or another SIP Trunk page.

Service Name - Defines a name for the service on this trunk.

Service Provider Domain - The FQDN or IP address of the ITSP SIP server. This domain name will be used in the Request-URI and To header field for outgoing SIP requests. If there are two redundant SIP Servers, you should enter both here, separated by comma. (Do not enter both SIP Servers, if these instead are addressed by DNS SRV records for the Outbound Proxy.) If the first SIP Server is out of service, the second will be tried. You should also enter these SIP Servers for monitoring at the SIP Services > VoIP Survival page to speed up failover. Here you may also add URI parameters such as ";user=phone" at the end of the domain.

Restrict to calls from - If specified, only calls originating from any of the specified networks will be handled as an incoming call to this SIP Trunk page. If not specified, a call may origin from anywhere as long as the called number matches any numbers configured here.

Outbound Proxy - Optional. Outbound SIP proxy for this trunk.

Use alias IP address - If IP aliases have been configured, you can select one of them for usage when communicating on the SIP Trunk.

Outbound Gateway - If an extra WAN interface has been configured for specific SIP traffic, you can select to use this WAN interface when communicating over this SIP trunk.

Signaling transport and **Port number** - The transport protocol and port number (optional) for SIP messages on the SIP trunk. Automatic means the transport protocol is determined by automatic means as defined in RFC 3263 based on "Outbound Proxy" or "Service Provider Domain" settings.

From header domain - This setting selects the domain name to use in the From header field for outgoing SIP requests on the SIP Trunk.

- Provider domain = Use domain name from "Service Provider Domain" setting.
- Enterprise domain = The domain name is kept as received from the caller.
- External IP address = The IP address of the network interface on which the request is sent.
- As entered = Use the manually entered "From domain" name. Here you may also add URI parameters such as ";user=phone" at the end of the domain.

Host name in Request-URI of incoming calls - The host part of Request-URI is usually one of this units IP addresses and this setting should then not be used. However, if the SIP Trunk Provider uses the host name as a "trunk ID", this setting has to be set to the trunk ID you get from your provider.

Local/Remote Trunk Group Parameters (RFC 4904) - Optional, with this setting you enable support for trunk group parameters (TGP) according to RFC 4904. Enter `";tgrp=the_trunk_group;trunk-context=the_trunk_context"` (replace the_trunk_group and the_trunk_context with whatever you have got from the SIP Trunk Provider). Remote TGP

refers to parameters placed in R-URI of egress calls to service provider and parameters received in Contact of ingress calls from service provider. Local TGP is the opposite, parameters found in R-URI of ingress calls from service provider and parameters placed in Contact of egress calls to service provider.

Used as - This section selection lets you choose how to use the parameters:

- Originating Trunk Group Parameters = Read remote TGP and/or place local TGP in Contact header.
- Destination Trunk Group Parameters = Read local TGP and/or place remote TGP in R-URI.
- Originating and Destination T.G.P = The TGP will be used for both of the above.

Reading TGP found in ingress calls means that the TGP found is matched to the configured values here. If they match the call will be forwarded using any matching line or the main trunk line. If the incoming call contains TGP that don't match the call will not be considered for forwarding by any line on this page.

Preserve Max-Forwards - Don't decrease Max-Forwards value of the SIP message as it passes through this trunk interface. Recommended setting is No as Max-Forwards helps detecting message loops resulting from bad configuration. But for interoperability with trunks using a very low Max-Forwards, this setting is required. If your calls fail with a "483 Too many hops" message and you don't think there is a message loop, this setting may help.

Relay media for remote users - Makes this unit relay the media so that the ITSP gets it from this unit, although the media endpoint may be located anywhere on the Internet. Solves interoperability problems with some ITSPs. Try this if you have problems with media (one or no way audio) when users are connected to your LAN from a remote location or calls are transferred externally.

Exactly one Via header - This unit has a built-in SIP proxy that adds a Via header to SIP requests. Some ITSP's do not expect the SIP request to have been routed by a SIP proxy and this setting is intended to solve such interoperability problems.

'gin' registration (RFC 6140) - Enables support for gin, see draft-ietf-martini-gin. The registration will then contain bulk number contact URI's and require that intermediate proxies and the registrar supports gin.

Hide Record-Route - For the same reasons as for the Via header, the Record-Route is a SIP proxy header field that can be hidden.

Show only one To tag - In case the SIP request is sent to a proxy that forks the call to multiple targets, this setting makes sure the caller becomes unaware of that several phones are ringing for the same call. This is also an interop setting for systems not expecting a SIP proxy in call path.

SIP 3xx redirection to provider domain - If domain in Contact header of 3xx responses should be modified to the domain of service provider.

SIP 3xx redirection to caller domain - If domain in Contact header of 3xx responses should be modified to the domain of the caller (From URI).

Route incoming based on - Defines where to look for the called number for matching an incoming call in the tables below. Request-URI is the default but some ITSPs put this information in To or P-Called-Party-ID headers.

Service Provider domain is trusted - Select this to enable usage of P-Asserted-Identity header field on the SIP Trunk as defined in RFC 3325. See setting for Identity in the tables below. Some trunks may use this field as caller-ID.

Use P-Preferred-Identity - Puts the identity found in the table below into a P-Preferred-Identity field instead of P-Asserted-Identity for outgoing calls.

Max simultaneous calls - Optional CAC (Call Admission Control). Number of calls supported on the SIP Trunk. If the limit is reached, new calls will not be tried, instead a busy response is sent back to the caller.

Max simultaneous per Trunk Line - Optional. This defines how many calls each line supports before it should be regarded as busy.

3.2 Number Routing, Registration and Authentication

Outgoing Calls to the trunk are sent to a specific SIP Trunking page via **Forward To** in the Dial Plan. The from header in an outgoing call is searched for a match in the **From**-columns.

Incoming Calls from the trunk are first scanned through the **Incoming Trunk Match** columns and only sent to the Dial Plan if no match is found.

Main Trunk Line

Used as the default if no match or an empty field is encountered in the tables below. This line should be configured for the main telephone number, and should contain the User ID and Password for the service if only one set is given. Registration shall be enabled on this line (only) if the service uses "implicit registration".

Notice: The From-field for outgoing calls must be empty on this line. The columns for incoming calls may be left empty on this line.

PBX Lines

This table is used for DID numbers that are mapped to the PBX.

SIP Lines

This table is used for DID numbers that are directly mapped to SIP clients registered to (or through) this device, i.e. not to phones that are connected to the PBX.

There are the following columns in the tables

No. - Just line number used for sorting the rows. Change numbers if you want to change the order of rows.

Reg - Set to Yes if this account should be registered at the ITSP SIP Server, e.g if implicit registration (or registration for all DID numbers) is used.

From Number/User - Can contain a number/name or regular expression. For outgoing calls, this field is matched to the calling user's user name in the From SIP URI (often a number but can also be a name). If there is a match, this line is selected for usage for the outgoing call. This field does not exist for Main Trunk Line as it is defined as the line to use if there is no match on any individual line.

Display Name - Optional SIP display name to use for outgoing calls, intended for presentation to the called party as a human readable string. In this field you can use the result of a subexpression from a match in a regular expression defined in the **From Number/User** field on the same row. E.g. dialing from 1306770713 having 1306(7707[0-9]{2}) in **From Number/User** with 0$1 in **Display Name**, will show 0770713 to the called party.

User Name - The SIP user name to use in the From SIP URI for outgoing calls and registrations. This is usually the telephone number of the ITSP SIP account and usually the number displayed as

caller ID on the PSTN. In this field you can use the result of a subexpression from a match in a regular expression defined in the **From Number/User** field on the same row. E.g. dialing from 1306770713 having 1306(7707[0-9]{2}) in **From Number/User** with +$0 in **User Name**, will result in +1306770713 in the From header to the ITSP. Generic Header Manipulation can be used in this field.

Identity - Optional value to use in P-Asserted-Identity or P-Preferred-Identity for outgoing calls. See "Use P-Preferred-Identity" above. The ITSP may take this as the caller ID to use on the PSTN. In this field you can use the result of a subexpression from a match in a regular expression defined in the **From Number/User** field on the same row. E.g. dialing from 1306770713 having 1306(7707[0-9]{2}) in **From Number/User** with +$0@goodservice.com in **Identity**, will result in +1306770713@goodservice.com in the P-Asserted- or P-Preferred-Identity header to the ITSP. Here you may also add URI parameters such as ";user=phone" at the end of the domain.

Authentication User ID and **Password** - Optional digest authentication user id and password if the trunk requires authentication. The Main Line settings will be used if empty in a matching line.

Incoming Trunk Match - The number or regular expression entered here has to match an incoming call to enable forwarding as specified in the **Forward to...** field.

Forward to... - For PBX Lines, this is the number an incoming call will be forwarded to on the PBX. For SIP Lines, this is the SIP address (complete SIP URI or a SIP user name/number on this unit's SIP Server) an incoming call will be forwarded to. In this field you can use the result of a subexpression from a match in a regular expression defined in the **Incoming Trunk Match** field on the same row. E.g. an incoming call to +1306770713 having \+1306(7707[0-9]{2}) in **Incoming Trunk Match** with 0$1 in **Forward to...** will result in forwarding to the PBX number 0770713. Here you may also add URI parameters such as ";user=phone" at the end of the domain of a complete URI. Generic Header Manipulation can be used in this field.

Information and help regarding Regular Expressions can be found here.

3.3 Setup for the PBX

This section contains settings for the interface towards the PBX.

Use PBX from other SIP Trunk / Define PBX settings - Determines whether the settings for the PBX should be taken from this page or from another SIP Trunk page.

PBX Name - Defines a name for the PBX on this page.

Use alias IP address - Optional. If IP aliases have been configured, you can select one of them for usage when communicating with the PBX.

PBX Registration SIP Address - Optional. If the PBX registers on this unit, this defines the SIP account (address-of-record) it registers to.

Authentication User ID and **Password** - Will be used for digest authentication of registration by the PBX, if entered. Will also be used for Digest Authentication of invites for outgoing calls, if enforced by the Dial Plan.

PBX IP address - The IP address of the PBX. Recommended to always enter if fixed.

PBX Domain Name - Optional SIP domain name of the PBX in case the PBX wants incoming calls be addressed to sip:number@domain instead of sip:number@ip-address. You shall also use this field if you have two redundant PBXs. Then add the IP address or FQDN of both PBXs in this field, separated by comma. (Do not fill in the PBX IP Address field then.) If the first PBX is out of service, the second will be tried. You should also enter these PBXs for monitoring at the SIP

Services > VoIP Survival page to speed up failover. Here you may also add URI parameters such as ";user=phone" at the end of the domain.

PBX Network - Specifies which "Network and Computers" the PBX is connected to.

Signaling transport and **Port number** - The transport protocol and port number (optional) for SIP messages sent to the PBX. Automatic means the transport protocol is determined automatically by applying the rules of RFC 3263 on the SIP URI sent to PBX.

Match From Number/User in field - Sets which field in the SIP message that will used as the callers number when matching the **From Number/User** column in the table specifications above.

To header field - Set the To header field to use for incoming calls to the PBX:

- Same as Request-URI = The To header field is set to the same value as the Request-URI.
- Copy from Trunk = Copies the To URI from the incoming call on the Trunk interface.
- Initial Request-URI = Set to equal the Request-URI as it looks initially before passing internal SIP proxy.
- As entered = Enter the URI to use in To header manually in the box to the right. Variable substitution as described above is available in this box. Here you may also add URI parameters such as ";user=phone" at the end of the domain.

Local/Remote Trunk Group Parameters usage - Optional setting for using the Trunk Group Parameters defined above also on the PBX interface.

- Originating Trunk Group Parameters = Read remote TGP and/or place local TGP in Contact header.
- Destination Trunk Group Parameters = Read local TGP and/or place remote TGP in R-URI.
- Originating and Destination T.G.P = The TGP will be used for both of the above.