Configuration Aid To Ingate Firewall/SIParator - SIP Trunking Configuration

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How To Use Your SIP Operator Account Via Ingate Firewall/SIParator

This is how to configure your firewall/SIParator to register at your SIP operator, and to use that SIP account for your local users.

This feature is only available when the Advanced SIP Routing or the SIP Trunking module has been installed.

Define a local SIP domain. This can be any domain name you like, as long as it isn't an existing domain somewhere else. A good choice is to use your company www domain, but replace the "www" with "sip", like *sip.ingate.com*. The same domain can also be used in pure SIP-to-SIP calls.

This domain should be entered on the **User Database** page under **SIP Traffic**.

Local SIP Domains (Help)

Edit	Domain	Delete
	sip.ingate.com	

Then, you define your local users in the **Local SIP User Database** table. These users will register on the firewall/SIParator with the usernames you enter here. Enter also their passwords and select a network from which they are allowed to register.

Note that no local user can have the same username as any of your operator account names.

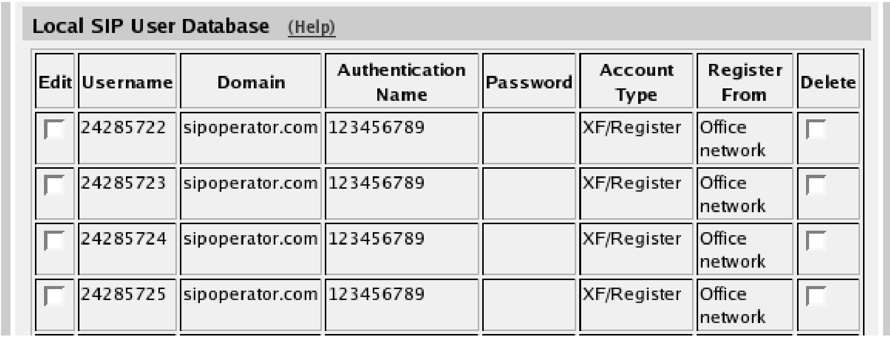
Local SIP User Database (Help)

Edit Row	Username	Domain	Authentication name	Password	Account type	Register from	Delete Row
	arthur	sip.ingate.com			User	All	
	harry	sip.ingate.com			User	All	
	helen	sip.ingate.com			User	All	
	mark	sip.ingate.com			User	All	
	test	sip.ingate.com			User	Office network	

Also enter your SIP operator account. You enter the username and password from the operator, and select the *XF/Register* account type. This account type will make the firewall/SIParator register at the SIP operator with the credentials you enter.

Some operators don't require registration. In this case, select the *XF* account type instead.

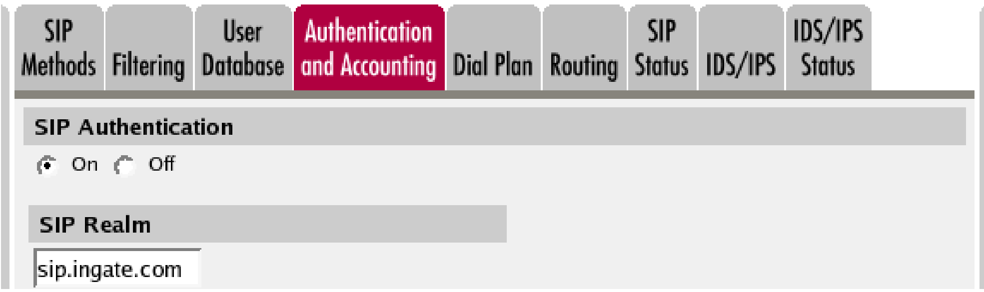
You can select any network in the Register from field, as it is not used for these account types.



Local SIP User Database (Help)

Edit	Username	Domain	Authentication Name	Password	Account Type	Register From	Delete
☐	24285722	sipoperator.com	123456789		XF/Register	Office network	☐
☐	24285723	sipoperator.com	123456789		XF/Register	Office network	☐
☐	24285724	sipoperator.com	123456789		XF/Register	Office network	☐
☐	24285725	sipoperator.com	123456789		XF/Register	Office network	☐

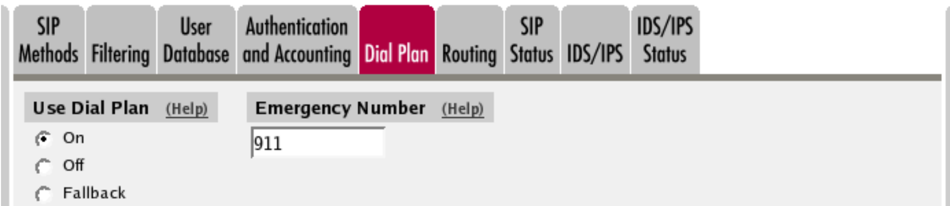
Go to the **Authentication and Accounting** page and turn authentication on. Also enter your SIP domain as the Realm.



Outgoing Calls

For outgoing calls, you have to define when your SIP operator account should be used. Usually, you use this type of account to call to the PSTN network ("ordinary telephones").

On the **Dial Plan** page, you define what type of calls should be redirected to your SIP operator. First, turn the Dial Plan on.



Show One Number When Calling

You can select to show one single calling number regardless of which user makes the call. This is useful when you want others to use your Answering service/Auto Attendant when calling back to you.

In the **Matching From Header** table, you define from which network the calls can come. You can also select what the From header (that tells who is calling) should look like. This is used when matching requests in the **Dial Plan** table below. Name each definition properly, to make it easier to use further on.

Matching From Header (Help)

Edit row	Name	Use this ...		... or this	Transport	Network	Delete row
		Username	Domain	Reg Exp			
☐	Any	*	*		Any	-	☐
☐	Office	*	*		TCP or TLS	Office network	☐

Add new rows [1] rows.

In the **Matching Request-URI** table, you define callees. This is used when matching requests in the **Dial Plan** table below.

In this case, you want to define the calls that should be routed to your SIP operator, which is call destinations where the usernames consist of numbers only, as these most likely are intended to go to the PSTN network. Call destinations that look like *helen@sip.ingate.com* should not be routed via the SIP operator, but be handled by the firewall/SIParator itself.

You can let users call international numbers with a + sign instead of the international prefix. For this, define the + sign as a **Prefix**, which means that it will be stripped before the call is forwarded.

The **Min. Tail** is set to 4 here, to open for the possibility of three-digit local extensions, which should not be handled by the **Dial Plan**.

Matching Request-URI (Help)

Edit Row	Name	Use this ...					... or this	Delete Row
		Prefix	Head	Tail	Min. Tail	Domain	Reg Exp	
☐	External numbers			0..9	4	*local		☐
☐	International numbers	+		0..9	4	*local		☐

In the **Forward To** table, you define where calls should be forwarded. This is used in the **Dial Plan** table below.

In this case, the calls should be forwarded to your SIP operator account that was defined before. You select the account under **Account**.

Forward To (Help)

Edit	Name	Subno.	Use This ...	... Or This			... Or This	Delete
			Account	Replacement URI	Port	Transport	Reg Expr	
☐	+ SIP Operator	1	24285722@sipoperator.com			-		☐

At last, you combine these definitions in the **Dial Plan** table. Make one line for international calls and one for other calls, because we need to add the international prefix for international calls only.

Dial Plan (Help)

Edit	No.	From Header	Request-URI	Action	Forward To	Add Prefix		ENUM Root	Time Class	Comment	Delete
						Forward	ENUM				
☐	1	Office	International numbers	Forward	SIP Operator	00		-	24/7	Change prefix for international calls.	☐
☐	2	Office	External numbers	Forward	SIP Operator			-	24/7	External calls sent to operator.	☐

Now, when a local user calls an external phone number, the firewall/SIParator will route this call to your SIP operator and rewrite the signaling to use your SIP operator account.

Show Different Numbers When Calling

You can select to show different calling numbers based on which user makes the call. This is useful when you want to let the called person use number presentation to see who is calling.

In the **Matching From Header** table, you define from which network the calls can come. You can also select what the From header (that tells who is calling) should look like. This is used when matching requests in the **Dial Plan** table below. Name each definition properly, to make it easier to use further on.

Create one row per user. These will be used to present the correct calling number for the called user.

Matching From Header (Help)

Edit	Name	Use This ...		... Or This	Transport	Network	Delete
		Username	Domain	Reg Expr			
☐	From Arthur	arthur	*local		Any	Office network	☐
☐	From Harry	harrry	*local		Any	Office network	☐
☐	From Helen	helen	*local		Any	Office network	☐
☐	From Mark	mark	*local		Any	Office network	☐

In the **Matching Request-URI** table, you define callees. This is used when matching requests in the **Dial Plan** table below.

In this case, you want to define the calls that should be routed to your SIP operator, which is call destinations where the usernames consist of numbers only, as these most likely are intended to go to the PSTN network. Call destinations that look like *helen@sip.ingate.com* should not be routed via the SIP operator, but be handled by the firewall/SIParator itself.

You can let users call international numbers with a + sign instead of the international prefix. For this, define the + sign as a **Prefix**, which means that it will be stripped before the call is forwarded.

The **Min. Tail** is set to 4 here, to open for the possibility of three-digit local extensions, which should not be handled by the **Dial Plan**.

Matching Request-URI (Help)

Edit Row	Name	Use this ...					... or this	Delete Row
		Prefix	Head	Tail	Min. Tail	Domain	Reg Exp	
☐	External numbers			0..9	4	*local		☐
☐	International numbers	+		0..9	4	*local		☐

In the **Forward To** table, you define where calls should be forwarded. This is used in the **Dial Plan** table below.

In this case, calls from one user should be forwarded to the corresponding SIP operator account. Create one row per user and select the account under **Account**.

Forward To (Help)

Edit	Name	Subno.	Use This ...	... Or This			... Or This	Delete
			Account	Replacement URI	Port	Transport	Reg Expr	
☐	+ Arthur PSTN	1	24285723@sipoperator.com			-		☐
☐	+ Harry PSTN	1	24285724@sipoperator.com			-		☐
☐	+ Helen PSTN	1	24285725@sipoperator.com			-		☐
☐	+ Mark PSTN	1	24285722@sipoperator.com			-		☐

At last, you combine these definitions in the **Dial Plan** table. For each user, make one line for international calls and one for other calls, because we need to add the international prefix for international calls only.

Dial Plan (Help)

Edit	No.	From Header	Request-URI	Action	Forward To	Add Prefix		ENUM Root	Time Class	Comment	Delete
						Forward	ENUM				
☐	1	From Helen	International numbers	Forward	Helen PSTN	00		-	24/7	Change prefix for international calls.	☐
☐	2	From Helen	External numbers	Forward	Helen PSTN			-	24/7	External calls sent to operator.	☐
☐	3	From Arthur	International numbers	Forward	Arthur PSTN			-	24/7		☐
☐	4	From Arthur	External numbers	Forward	Arthur PSTN			-	24/7		☐
☐	5	From Harry	International numbers	Forward	Harry PSTN			-	24/7		☐
☐	6	From Harry	External numbers	Forward	Harry PSTN			-	24/7		☐
☐	7	From Mark	International numbers	Forward	Mark PSTN			-	24/7		☐
☐	8	From Mark	External numbers	Forward	Mark PSTN			-	24/7		☐

Now, when a local user calls an external phone number, the firewall/SIParator will route this call to your SIP operator and rewrite the signaling to use your SIP operator account.

Incoming Calls

If your SIP account provides several phone numbers, you can assign separate numbers for your local users. You do that on the **Routing** page.

There are two different ways of mapping phone numbers to users; either the PSTN numbers are mapped to users or the users are given numbers as aliases. The latter only works when the Advanced SIP Routing module has been installed and the SIP operator does not require registration.

In the **User Routing** table, you can select each phone number, and enter which user calls should be forwarded to.

User Routing (Help)

Edit	User	Alias	Restrict Incoming Callers	Forward		Send To Voice Mail	Time Class	Comment	Delete
				Action	To				
	24285722@sipoperator.com		Off	Forward	mark@sip.ingate.com	-	-		
	24285723@sipoperator.com		Off	Forward	arthur@sip.ingate.com	-	-		
	24285724@sipoperator.com		Off	Forward	harry@sip.ingate.com	-	-		
	24285725@sipoperator.com		Off	Forward	helen@sip.ingate.com	-	-		

You can also select a local user and assign a SIP operator phone number as an Alias for that user. This will only work when the Advanced SIP Routing module has been installed and the SIP operator does not require registration.

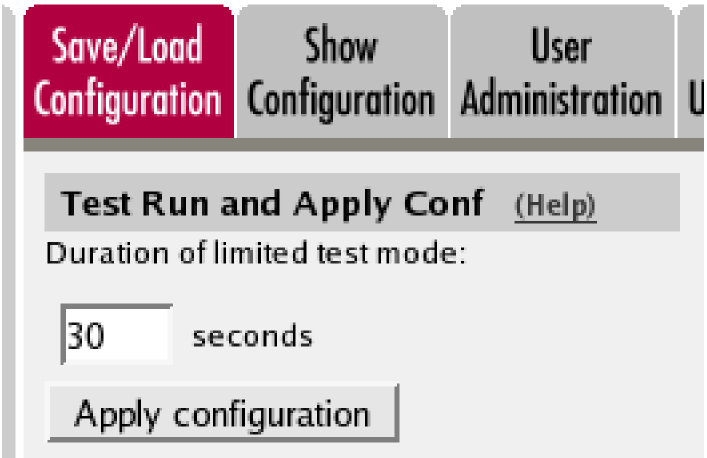
User Routing (Help)

Edit	User	Alias	Restrict Incoming Callers	Forward		Send To Voice Mail	Time Class	Comment	Delete
				Action	To				
	arthur@sip.ingate.com	24285723	Off	-		-	-		
	harry@sip.ingate.com	24285724	Off	-		-	-		
	helen@sip.ingate.com	24285725	Off	-		-	-		
	mark@sip.ingate.com	24285722	Off	-		-	-		

Now, when someone calls 34382753, the call will be routed from the SIP operator to the firewall/SIParator and finally to *harry@sip.ingate.com*.

Note that you can only use the **User Routing** table for incoming call forwarding. The **Static Registrations** should not be used when XF or XF/Register accounts are involved.

Finally, go to the **Save/Load Configuration** page under **Administration** and apply the new settings by pressing **Apply configuration**.


Save/Load Configuration
Show Configuration
User Administration
Test Run and Apply Conf
(Help)
Duration of limited test mode:
30
seconds
Apply configuration