



FIBERME Communications LLC.

Configuring FAG410X with FreePBX

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OVERVIEW

This document describes basic configuration to interconnect FreePBX and FAG410X. In this document, we are using FAG4108 as an example. The following methodology can be used for the FAG4104 as well. This is typically applied to the scenario where users would like to add a FAG410X not only as a remote extension but also as an external PSTN trunk.

There are two ways to set up the FreePBX with the FAG410X.

- **Method 1:** Configure FAG410X as a SIP Peer Trunk.
- **Method 2:** Register FAG410X on FreePBX directly as an extension.



CONNECT FreePBX TO FAG410X USING PEER SIP TRUNK

Create IVR On FreePBX

On the FreePBX web GUI, create an IVR extension under **Applications** ->**IVR** ->**Add IVR**.

In IVR settings, if "Direct Dial" is enabled, the calls dialing into the FreePBX IVR will be able to reach the internal extensions registered to FreePBX. Also, you can assign the "IVR Entries" to different destinations.

The screenshot shows the 'Add IVR' configuration page in the FreePBX web GUI. The page is titled 'Add IVR' and contains several sections: 'IVR General Options', 'IVR DTMF Options', and 'IVR Entries'. The 'IVR General Options' section includes fields for 'IVR Name' (FAG410X) and 'IVR Description'. The 'IVR DTMF Options' section includes various settings like 'Announcement', 'Enable Direct Dial', 'Force Strict Dial Timeout', 'Timeout', 'Alert Info', 'Ringer Volume Override', 'Invalid Retries', 'Invalid Retry Recording', 'Append Announcement to Invalid', 'Return on Invalid', 'Invalid Recording', 'Invalid Destination', 'Timeout Retries', 'Timeout Retry Recording', 'Append Announcement on Timeout', 'Return on Timeout', 'Timeout Recording', 'Timeout Destination', and 'Return to IVR after VM'. The 'IVR Entries' section is a table with columns for 'Digits', 'Destination', 'Return', and 'Delete'. It contains two entries: '0' with destination 'Queues' and '4400' with destination 'Recursion'. The 'Return' column has 'Yes' and 'No' buttons. The 'Delete' column has a trash icon. At the bottom right, there are 'Submit', 'Duplicate', and 'Reset' buttons.

Figure 1: Create IVR on FreePBX



Create Peer SIP TRUNK On FreePBX

On the FreePBX web GUI, create a peer SIP trunk under **Connectivity** ->**Trunks** ->**Add SIP (Chan_pjsip) Trunk**. In this example, the FAG4108 IP address is 192.168.99.239.

Add Trunk

The screenshot displays the 'Add Trunk' configuration interface in the FreePBX web GUI. It is divided into two main sections: 'General' and 'PJSIP Settings'.

General Section:

- Trunk Name:** FAG410X
- Hide CallerID:** Yes (selected), No
- Outbound CallerID:** (empty field)
- CID Options:** Allow Any CID (selected), Block Foreign CIDs, Remove CNAM, Force Trunk CID
- Maximum Channels:** (empty field)
- Asterisk Trunk Dial Options:** T (selected), Override, System
- Continue if Busy:** Yes (selected), No
- Disable Trunk:** Yes (selected), No
- Monitor Trunk Failures:** Yes (selected), No

PJSIP Settings Section:

- General Tab (selected):**
 - Username:** Authentication Disabled
 - Auth username:** Authentication Disabled
 - Secret:** Authentication Disabled
 - Authentication:** Outbound (selected), Inbound, Both, None
 - Registration:** Send (selected), Receive, None
 - Language Code:** Default
 - SIP Server:** 192.168.99.239
 - SIP Server Port:** 5060
 - Context:** from-pstn
 - Transport:** 0.0.0.0-udp
- Advanced Tab:** (empty)
- Codecs Tab:** (empty)

At the bottom right of the PJSIP Settings section, there are 'Submit' and 'Reset' buttons.

Figure 2: Create Peer SIP Trunk on FreePBX

Configure Outbound Rule on FreePBX

On FreePBX web GUI, go to **Connectivity** ->**Outbound Routes** ->**Add Outbound Route** to create a new outbound rule. This would allow the extension on FreePBX to reach numbers in PSTN network via the peer SIP trunk we just configured.



Outbound Routes

Add Route

Route Settings	Dial Patterns	Import/Export Patterns	Notifications	Additional Settings
Route Name FAG410X_Outbound				
Route CID				
Override Extension Yes No				
Route Password				
Route Type Emergency Intra-Company				
Music On Hold? default				
Time Match Time Zone: Use System Timezone				
Time Match Time Group ---Permanent Route---				
Trunk Sequence for Matched Routes				
+ FAG410X				
+				
Route Settings	Dial Patterns	Import/Export Patterns	Notifications	Additional Settings
Dial Patterns that will use this Route				
Pattern Help +				
Dial patterns wizards				
(prepend) 9 [X.] / CallerID +				

Figure 3: Configure Outbound Rule on FreePBX

In this example prefix "9", 9 is the first dialing digit and it will be stripped off when the call goes out.

Configure Inbound Rule on FreePBX

On FreePBX web GUI, go to **Connectivity -> Inbound Routes -> Add Inbound Route** to create a new inbound rule. In this example, we set the DID Number as **20000**, which will be used in the FAG410X call forward setting.

Inbound Routes

Add Incoming Route

General	Advanced	Privacy	Fax	Other
Description FAG410X_Inbound				
DID Number 20000				
CallerID Number ANY				
CID Priority Route Yes No				
Alert Info None				
Ringer Volume Override None				
CID name prefix				
Music On Hold Default				
Set Destination IVR				
FAG410X				
» Submit Reset				

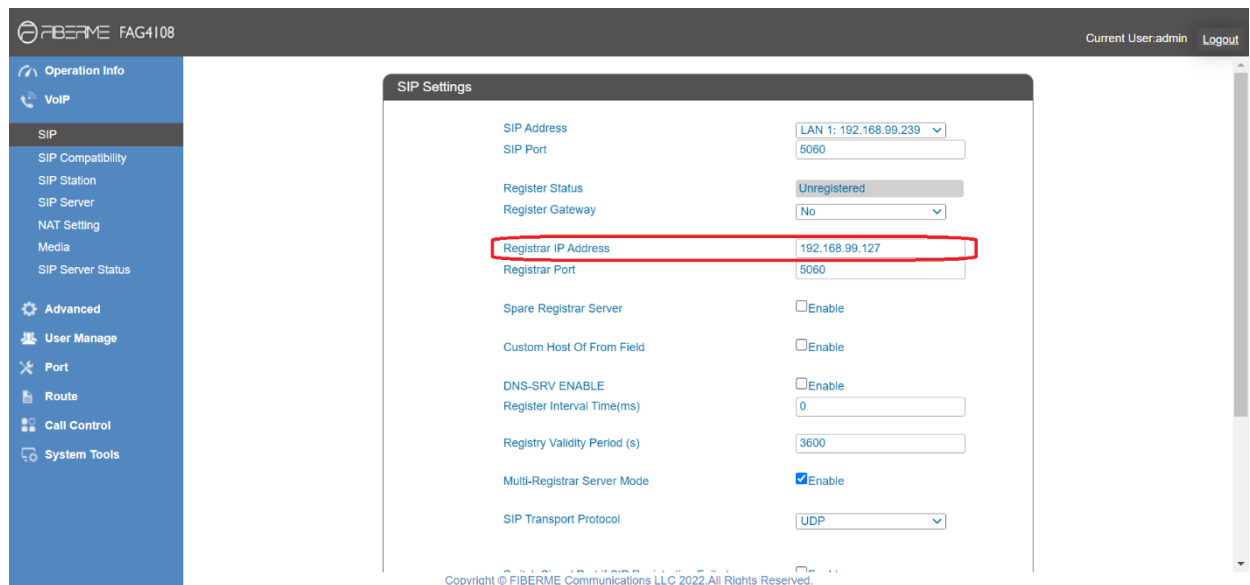
Figure 4: Configure Inbound Rule on FreePBX

The default destination is configured to IVR. Ensure to select the proper IVR.



Connect FAG410X with FreePBX

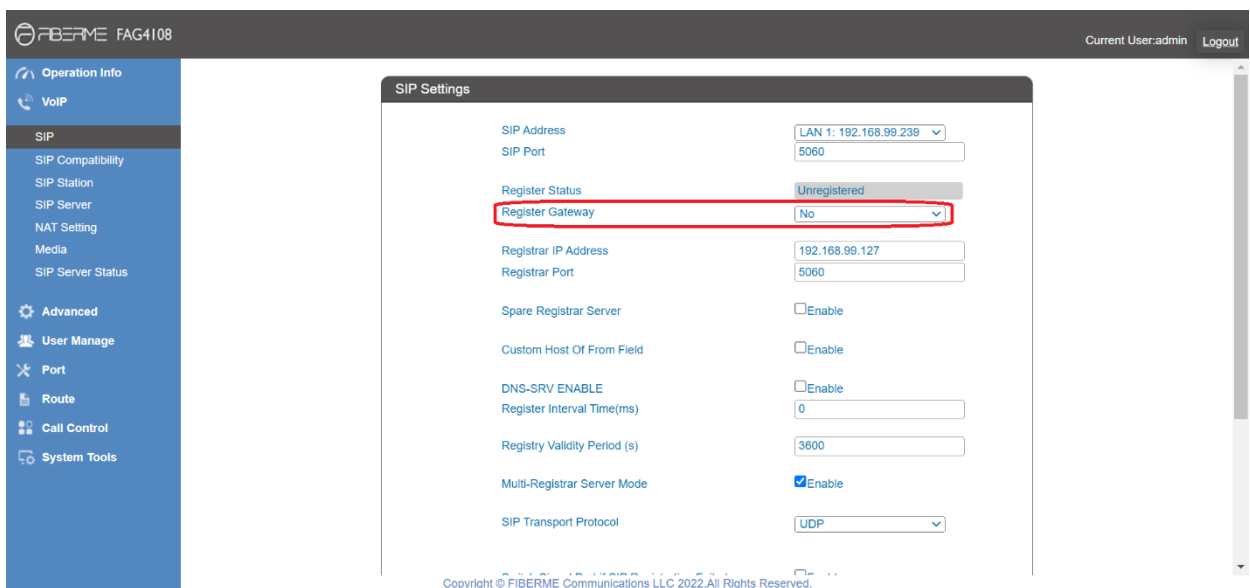
1. On the FAG410X web GUI, go to the **VoIP->SIP** page and enter the IP address of the FreePBX that you are peering with.



The screenshot shows the FAG410X web GUI with the 'SIP Settings' page. The left sidebar contains a menu with 'VoIP' selected, and 'SIP' is highlighted under the 'VoIP' section. The main content area displays various SIP settings. The 'Registrar IP Address' field is highlighted with a red box and contains the value '192.168.99.127'. Other settings include 'SIP Address' (LAN 1: 192.168.99.239), 'SIP Port' (5060), 'Register Status' (Unregistered), 'Register Gateway' (No), 'Spare Registrar Server' (disabled), 'Custom Host Of From Field' (disabled), 'DNS-SRV ENABLE' (disabled), 'Register Interval Time(ms)' (0), 'Registry Validity Period (s)' (3600), 'Multi-Registrar Server Mode' (checked), and 'SIP Transport Protocol' (UDP).

Figure 5: Connect FAG410X with FreePBX: Registrar IP Address

2. Please make sure the **Register Gateway** option under **VoIP-> SIP** is set to **No**. In the following example, FreePBX Server has IP address 192.168.99.127.



The screenshot shows the FAG410X web GUI with the 'SIP Settings' page. The left sidebar contains a menu with 'VoIP' selected, and 'SIP' is highlighted under the 'VoIP' section. The main content area displays various SIP settings. The 'Register Gateway' field is highlighted with a red box and contains the value 'No'. Other settings include 'SIP Address' (LAN 1: 192.168.99.239), 'SIP Port' (5060), 'Register Status' (Unregistered), 'Registrar IP Address' (192.168.99.127), 'Register Port' (5060), 'Spare Registrar Server' (disabled), 'Custom Host Of From Field' (disabled), 'DNS-SRV ENABLE' (disabled), 'Register Interval Time(ms)' (0), 'Registry Validity Period (s)' (3600), 'Multi-Registrar Server Mode' (checked), and 'SIP Transport Protocol' (UDP).

Figure 6: Connect FAG410X with FreePBX: Register Gateway



Configure FXO Port on FAG410X

1. Connect the PSTN line to the FAG410X FXO port.
2. On the FAG410X web GUI, go to the **Port->FXO** page and press Modify on the FXO port you will use.

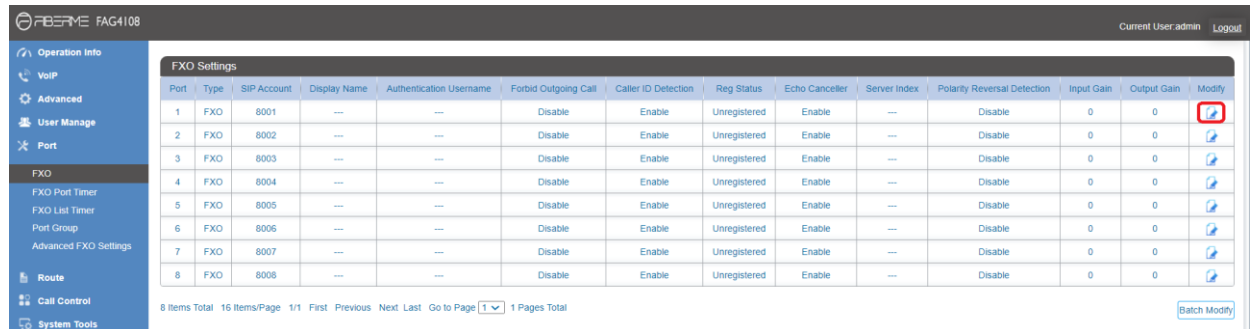


Figure 7 shows the 'FXO Settings' table in the FAG410X web GUI. The table lists 8 FXO ports. The 'Modify' button for Port 1 is highlighted with a red circle.









Port	Type	SIP Account	Display Name	Authentication Username	Forbid Outgoing Call	Caller ID Detection	Reg Status	Echo Cancellation	Server Index	Polarity Reversal Detection	Input Gain	Output Gain	Modify
1	FXO	8001	---	---	Disable	Enable	Unregistered	Enable	---	Disable	0	0	
2	FXO	8002	---	---	Disable	Enable	Unregistered	Enable	---	Disable	0	0	
3	FXO	8003	---	---	Disable	Enable	Unregistered	Enable	---	Disable	0	0	
4	FXO	8004	---	---	Disable	Enable	Unregistered	Enable	---	Disable	0	0	
5	FXO	8005	---	---	Disable	Enable	Unregistered	Enable	---	Disable	0	0	
6	FXO	8006	---	---	Disable	Enable	Unregistered	Enable	---	Disable	0	0	
7	FXO	8007	---	---	Disable	Enable	Unregistered	Enable	---	Disable	0	0	
8	FXO	8008	---	---	Disable	Enable	Unregistered	Enable	---	Disable	0	0	

Figure 7: Configure FXO Port on FAG410X: Modify Port

3. Enter the SIP Account Number **20000** to be matched with the DID number we configured in FreePBX Inbound Route.

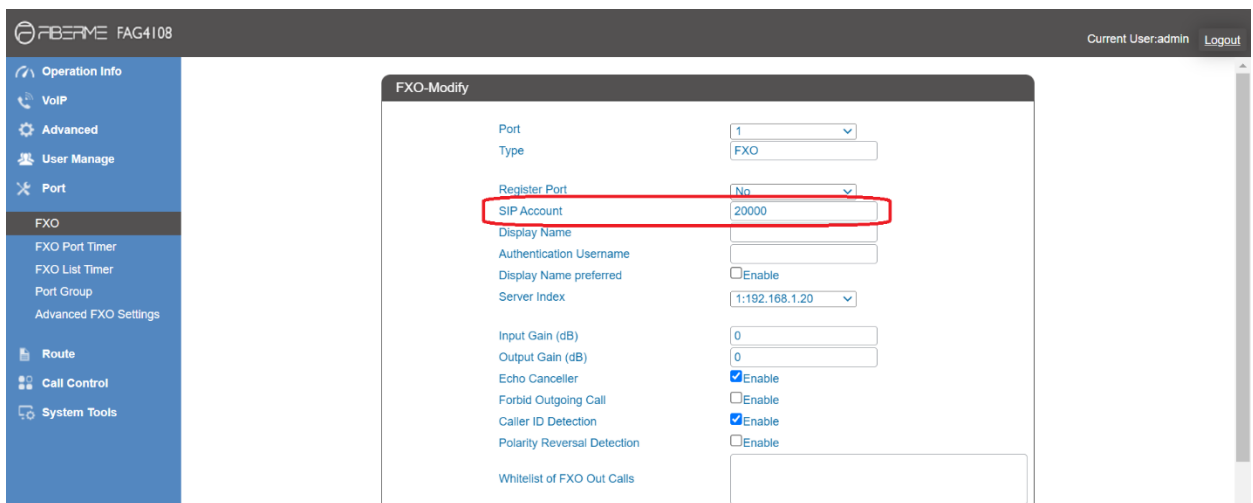


Figure 8 shows the 'FXO-Modify' form in the FAG410X web GUI. The 'SIP Account' field is highlighted with a red circle.

Field	Value
Port	1
Type	FXO
Register Port	No
SIP Account	20000
Display Name	
Authentication Username	
Display Name preferred	<input type="checkbox"/> Enable
Server Index	1:192.168.1.20
Input Gain (dB)	0
Output Gain (dB)	0
Echo Cancellation	<input checked="" type="checkbox"/> Enable
Forbid Outgoing Call	<input type="checkbox"/> Enable
Caller ID Detection	<input checked="" type="checkbox"/> Enable
Polarity Reversal Detection	<input type="checkbox"/> Enable
Whitelist of FXO Out Calls	

Figure 8: Configure FXO Port on FAG410X: Port Configurations



Create Port Group on FAG410X

1. On the FAG410X web GUI, go to the **Port->Port Group** page and press “Add New”.

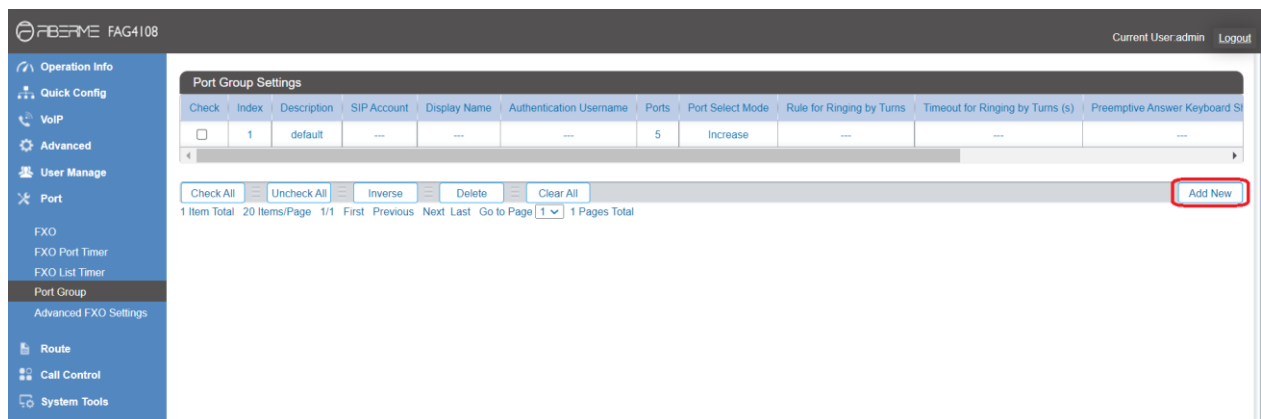


Figure 9: Create Port Group on FAG410X: Add New

2. Select the FXO port you will use.

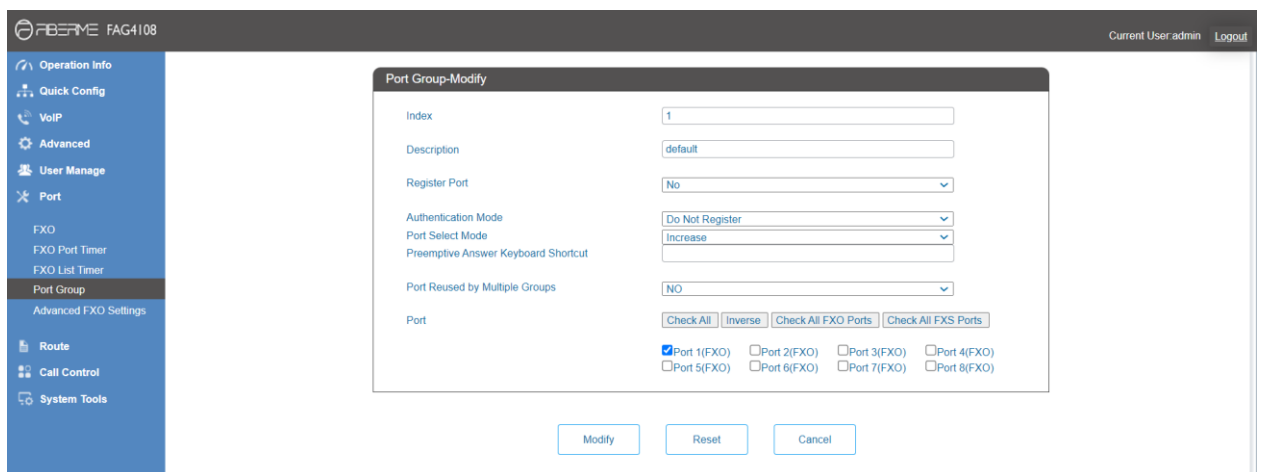


Figure 10: Create Port Group on FAG410X: Group Configurations



REGISTER FAG410X ON FreePBX AS AN EXTENSION

Create SIP Extension on FreePBX

To manually create new SIP user, go to FreePBX web GUI **Applications ->Extensions ->Add New SIP [chan_pjsip] Extension**. A new dialog window will show for users to fill in the extension information.

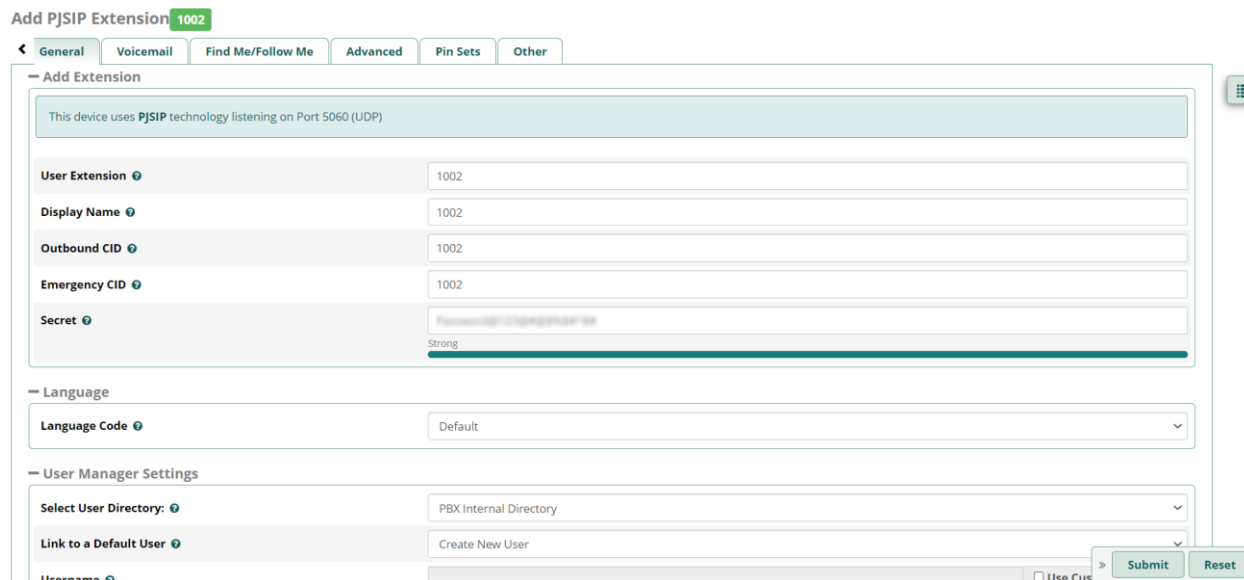


Figure 11: Create SIP Extension on FreePBX

Configure FAG410X SIP Account as a Registered Extension on FreePBX

Under FAG410X **web GUI->VoIP->SIP**, please set Register Gateway to Yes, then enter SIP Extension information created earlier in FreePBX. In this example, extension 1002 is used in order to register FAG410X as an extension user on FreePBX.

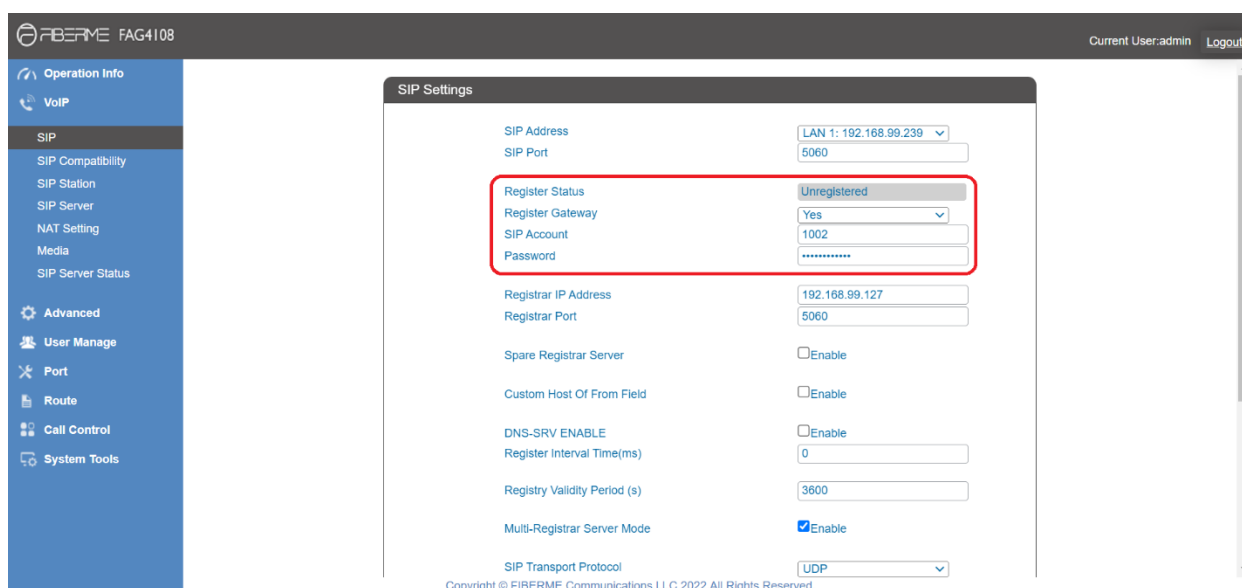
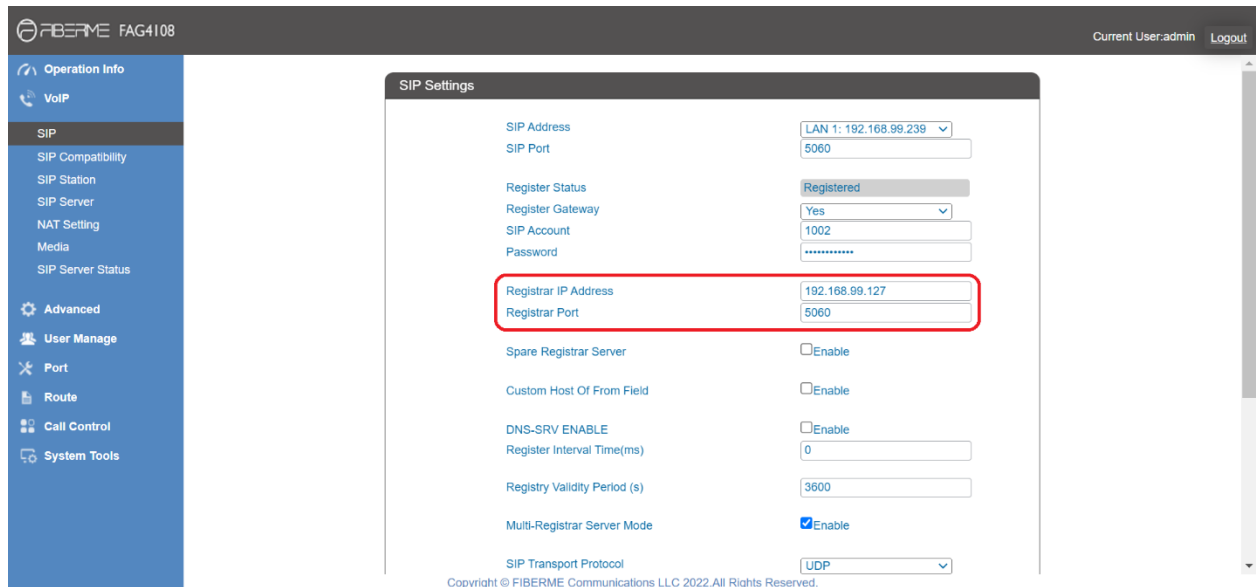


Figure 12: FAG410X SIP Account Settings



Under FAG410X web GUI, **VoIP->SIP**, please fill in FreePBX information as explained in method 1.

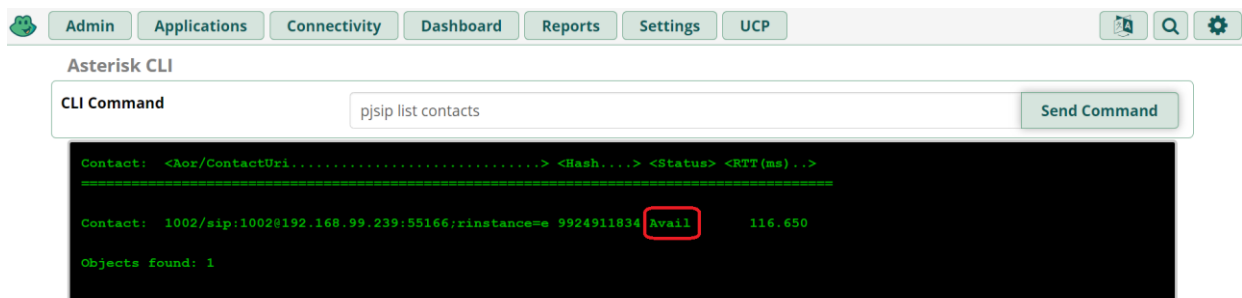


The screenshot shows the FAG410X web GUI with the 'SIP Settings' tab selected. The 'Registrar IP Address' field is highlighted with a red box. The settings are as follows:

Field	Value
SIP Address	LAN 1: 192.168.99.239
SIP Port	5060
Register Status	Registered
Register Gateway	Yes
SIP Account	1002
Password	*****
Registrar IP Address	192.168.99.127
Registrar Port	5060
Spare Registrar Server	<input type="checkbox"/> Enable
Custom Host Of From Field	<input type="checkbox"/> Enable
DNS-SRV ENABLE	<input type="checkbox"/> Enable
Register Interval Time(ms)	0
Registry Validity Period (s)	3600
Multi-Registrar Server Mode	<input checked="" type="checkbox"/> Enable
SIP Transport Protocol	UDP

Figure 13: FAG410X Registrar IP Address

We can check FreePBX SIP Extension Status to see if FAG410X has been successfully registered as an extension device. go to **Admin -> Asterisk-CLI**, In CLI Command write "pjsip list contacts" and press Send Command. "Avail" under Status means that the extension is registered successfully as below screenshot.



The screenshot shows the Asterisk CLI interface with the command 'pjsip list contacts' entered. The output shows the status 'Avail' highlighted with a red box.

```
CLI Command: pjsip list contacts
Contact: <Aor/ContactUri> <Hash> <Status> <RTT(ms)>
Contact: 1002/sip:1002@192.168.99.239:55166;rinstance=e 9924911834 Avail 116.650
Objects found: 1
```

Figure 14: FreePBX - SIP Extension Status

Now FAG410X is registered at FreePBX as an extension device. Please refer to method 1 in the previous section to configure FXO Port settings on FAG410X.



FAG410X CALL Routing

Configure IP to Tel on FAG410X

1. On the FAG410X web GUI, go to the **Route-> IP→TEL** page, Press “Add New”.

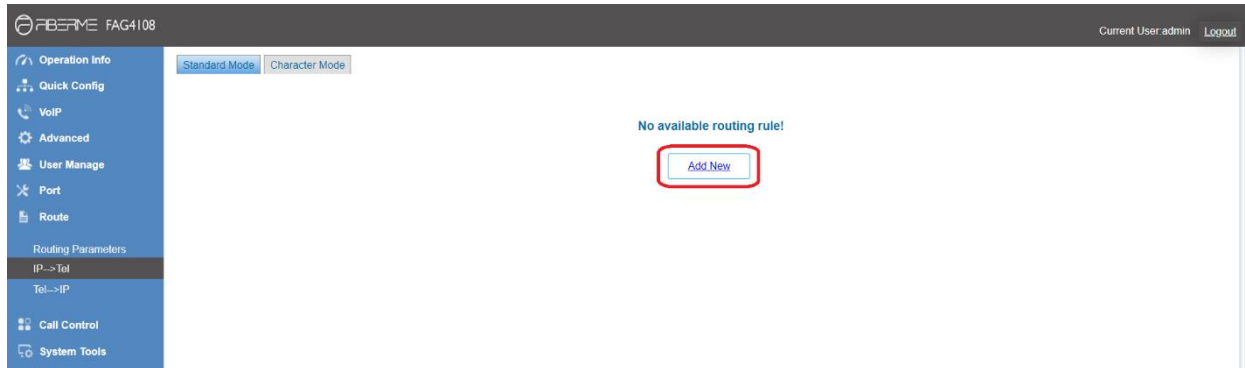


Figure 15: FAG410X – IP to TEL

2. Enter the FreePBX Server IP address in “Source IP” and select the Port Group you will use from “Call Destination”.

Figure 16: IP to TEL Settings



Configure TEL to IP on FAG410X

1. On the FAG410X web GUI, go to the **Route-> TEL->IP** page, Press “Add New”.

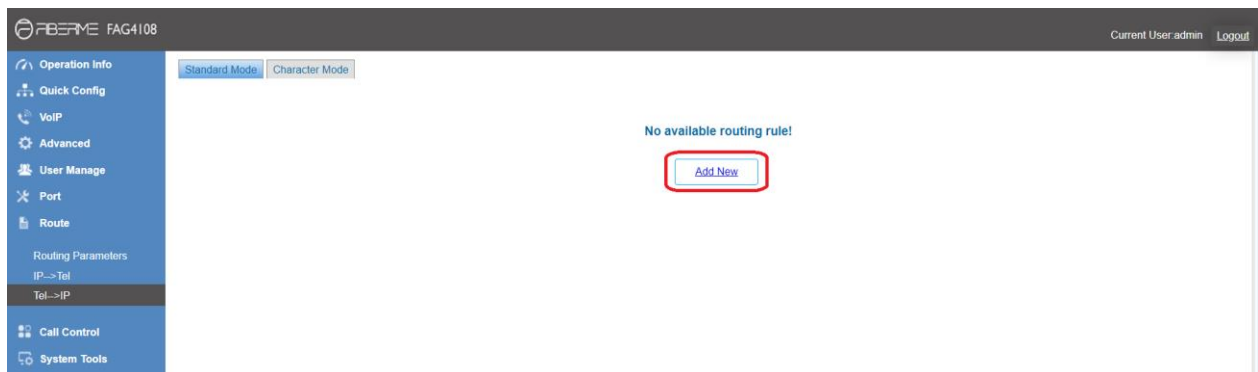


Figure 17: FAG410X – TEL to IP

2. Select the Port Group you will use from “Source Port Group” and enter the FreePBX Server IP address in “Destination Address”.

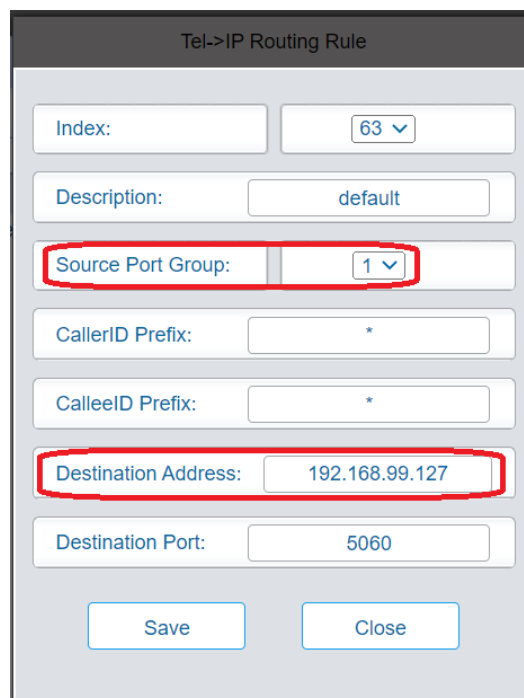


Figure 18: TEL to IP Settings

