



FIBERME Communications LLC.

Configuring FEG4301 with FreePBX

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OVERVIEW

This document describes basic configuration to interconnect FreePBX and FEG4301. In this document, we are using FEG4301 as an example. This is typically applied to the scenario where users would like to add a FEG4301 as an external ISDN trunk.



CONNECT FreePBX TO FEG4301 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 the FreePBX. Also, you can assign the "IVR Entries" to different destinations.

Dashboard Reports Settings UCP

Add IVR

IVR General Options

IVR Name: FEG4301_IVR

IVR Description:

IVR DTMF Options

Announcement: FEG4301

Enable Direct Dial: Disabled

Force Strict Dial Timeout: Yes No No - Legacy

Timeout: 10

Alert Info: None

Ringer Volume Override: None

Invalid Retries: 3

Invalid Retry Recording: Default

Append Announcement to Invalid: Yes No

Return on Invalid: Yes No

Invalid Recording: Default

Invalid Destination: None

Timeout Retries: 3

Timeout Retry Recording: Default

Append Announcement on Timeout: Yes No

Return on Timeout: Yes No

Timeout Recording: Default

Timeout Destination: None

Return to IVR after VM: Yes No

IVR Entries

Digits	Destination	Return	Delete
0	Queues		
	4400 Reception		

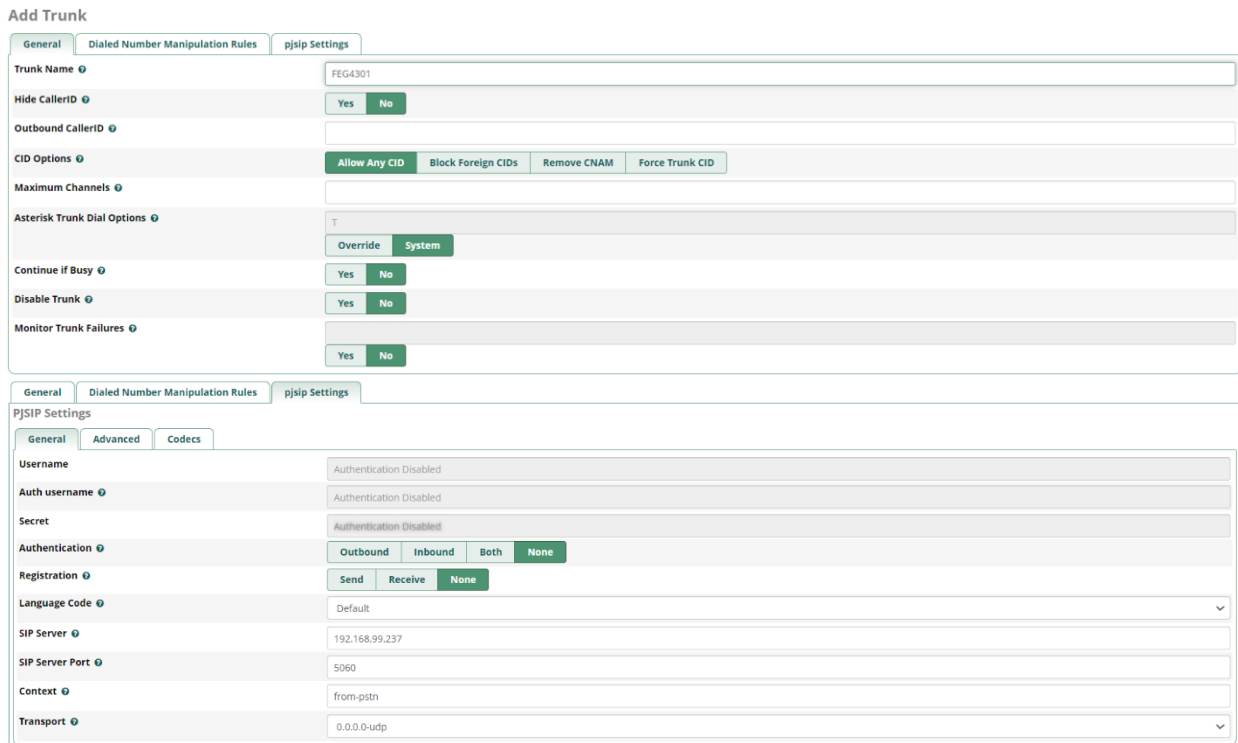
+Add Another Entry

Figure 1: Create IVR on the 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 FEG4301 IP address is 192.168.99.237.



Add Trunk

General | Dialed Number Manipulation Rules | pjsip Settings

Trunk Name

Hide CallerID Yes No

Outbound CallerID

CID Options Allow Any CID Block Foreign CIDs Remove CNAM Force Trunk CID

Maximum Channels

Asterisk Trunk Dial Options

Override System

Continue if Busy Yes No

Disable Trunk Yes No

Monitor Trunk Failures

Yes No

PJSIP Settings

General | Advanced | Codecs

Username

Auth username

Secret

Authentication Outbound Inbound Both None

Registration Send Receive None

Language Code

SIP Server

SIP Server Port

Context

Transport

Figure 2: Create Peer SIP Trunk on the FreePBX

Configure Outbound Rule on FreePBX

On the FreePBX web GUI, go to **Connectivity ->Outbound Routes ->Add Outbound Route** to create a new outbound rule. This would allow the extension on the FreePBX to reach numbers in ISDN 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

Route CID

Override Extension Yes No

Route Password

Route Type Emergency Intra-Company

Music On Hold?

Time Match Time Zone:

Time Match Time Group

Trunk Sequence for Matched Routes

-
-

Optional Destination on Congestion

Note: Extension Routes is not registered

Dial Patterns that will use this Route

Pattern Help

Figure 3: Configure Outbound Rule on the 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 the FreePBX web GUI, go to **Connectivity ->Inbound Routes ->Add Inbound Route** to create a new inbound rule. Let's say that we have 30 numbers on the PRI starting from 1122334450 to 1122334479. In this example, we create the DID as **11223344XX** to receive calls through all the 30 numbers.

Inbound Routes

Add Incoming Route

General | Advanced | Privacy | Fax | Other

Description

DID Number

CallerID Number

CID Priority Route Yes No

Alert Info

Ring Volume Override

CID name prefix

Music On Hold

Set Destination

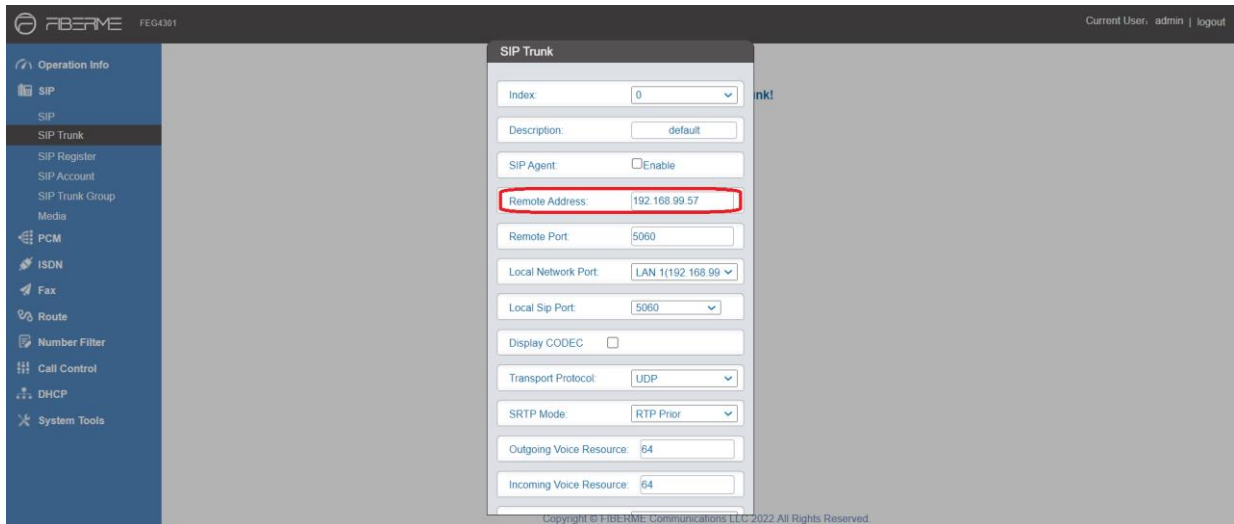
Figure 4: Configure Inbound Rule on FreePBX

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



Connect FEG4301 with FreePBX

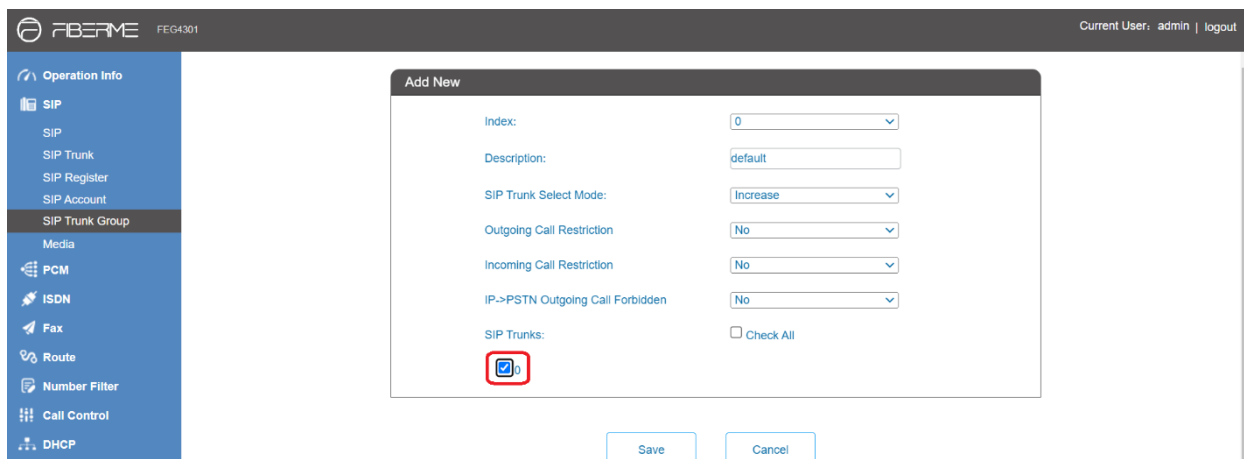
1. On the FEG4301 web GUI, go to the **VoIP->SIP Trunk** page and click on **Add New** button. Then, enter the IP address of the FreePBX PBX that you are peering with in Remote Address. In the following example, FreePBX has IP address 192.168.99.57.



The screenshot shows the FEG4301 web GUI interface. On the left is a navigation menu with categories like Operation Info, SIP, SIP Trunk, SIP Register, SIP Account, SIP Trunk Group, Media, PCM, ISDN, Fax, Route, Number Filter, Call Control, DHCP, and System Tools. The main content area displays the 'SIP Trunk' configuration form. The 'Remote Address' field is highlighted with a red rectangular box and contains the value '192.168.99.57'. Other fields include Index (0), Description (default), SIP Agent (unchecked), Remote Port (5060), Local Network Port (LAN 1(192.168.99)), Local Sip Port (5060), Display CODEC (unchecked), Transport Protocol (UDP), SRTP Mode (RTP Prior), Outgoing Voice Resource (64), and Incoming Voice Resource (64). The top right corner shows 'Current User: admin | logout' and the bottom center has a copyright notice for FIBERME Communications LLC 2022.

Figure 5: Connect FEG4301 with FreePBX: Remote Address

2. Go to the **VoIP->SIP Trunk Group** page and click on **Add New** button. Then check on the trunk we've created.



The screenshot shows the FEG4301 web GUI with the 'Add New' dialog box open. The dialog box contains the following fields: Index (0), Description (default), SIP Trunk Select Mode (Increase), Outgoing Call Restriction (No), Incoming Call Restriction (No), IP->PSTN Outgoing Call Forbidden (No), and SIP Trunks (a list with a checkbox for '0' checked and highlighted with a red box, and a 'Check All' checkbox). At the bottom of the dialog are 'Save' and 'Cancel' buttons. The background shows the same navigation menu as in Figure 5.

Figure 6: Connect FEG4301 with FreePBX: SIP Trunk Group



Create PCM Trunk Group on FEG4301

1. On the FEG4301 web GUI, go to the **PCM->PCM Trunk Group** page and press “Add New”. Then, check mark on the PCM trunk.

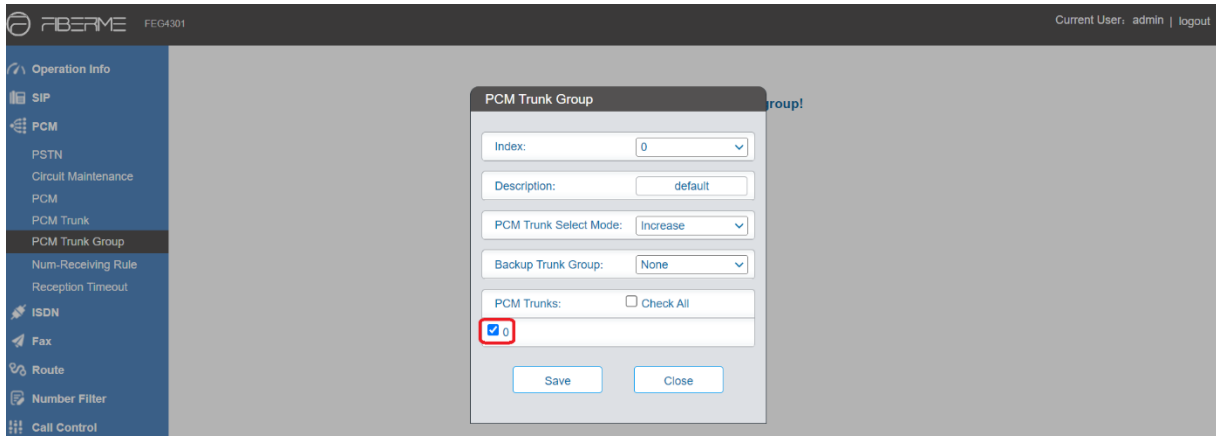


Figure 7: Create PCM Trunk Group on FEG4301



FEG4301 CALL Routing

Configure IP to PSTN on FEG4301

1. On the FEG4301 web GUI, go to the **Route-> IP->PSTN** page, Press “Add New”.

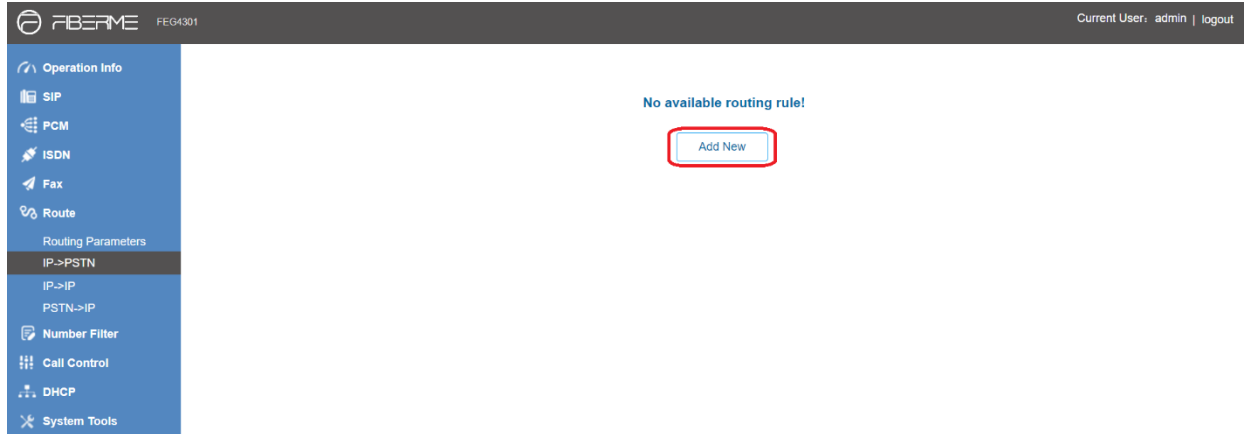


Figure 8: FEG4301 – IP to PSTN

2. Select the SIP Trunk Group in “Call Source” and the PCM Trunk Group in “Call Destination”.

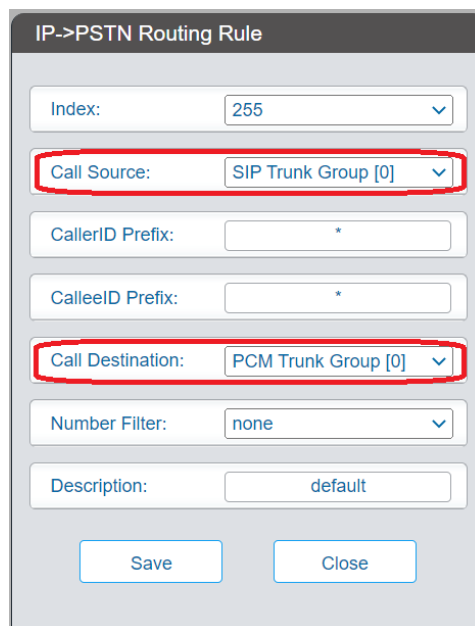
The screenshot shows a dialog box titled 'IP->PSTN Routing Rule'. It contains several input fields: 'Index' with a dropdown menu showing '255'; 'Call Source' with a dropdown menu showing 'SIP Trunk Group [0]'; 'CallerID Prefix' with a text input field containing '*'; 'CalleeID Prefix' with a text input field containing '*'; 'Call Destination' with a dropdown menu showing 'PCM Trunk Group [0]'; 'Number Filter' with a dropdown menu showing 'none'; and 'Description' with a text input field containing 'default'. At the bottom of the dialog are two buttons: 'Save' and 'Close'. The 'Call Source' and 'Call Destination' dropdown menus are highlighted with red rectangular boxes.

Figure 9: IP to PSTN Settings



Configure PSTN to IP on FEG4301

1. On the FEG4301 web GUI, go to the **Route-> PSTN->IP** page, Press “Add New”.

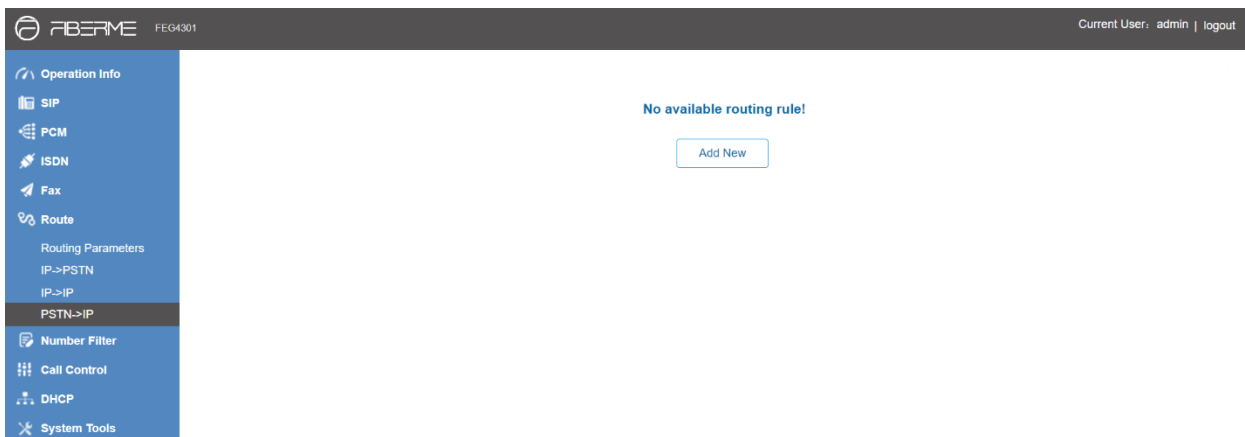


Figure 10: FEG4301 – PSTN to IP

2. Select the PCM Trunk Group in “Call Initiator” and the SIP Trunk Group in “Call Destination”.

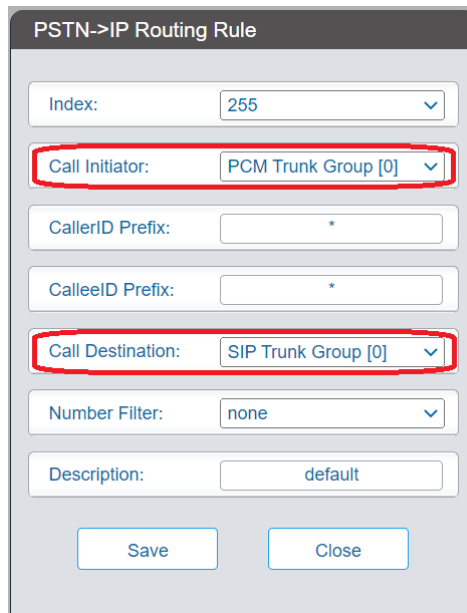
The image shows a 'PSTN->IP Routing Rule' configuration window. It contains several fields: 'Index' (255), 'Call Initiator' (PCM Trunk Group [0]), 'CallerID Prefix' (*), 'CalleeID Prefix' (*), 'Call Destination' (SIP Trunk Group [0]), 'Number Filter' (none), and 'Description' (default). The 'Call Initiator' and 'Call Destination' fields are highlighted with red rectangular boxes. At the bottom, there are 'Save' and 'Close' buttons.

Figure 11: PSTN to IP Settings

