



FIBERME Communications LLC.

Configuring FEG4301 with Elastix

Table of Contents

OVERVIEW.....	3
CONNECT Elastix TO FEG4301 USING PEER SIP TRUNK	4
Create IVR On Elastix	4
Create Peer SIP TRUNK On Elastix	5
Configure Outbound Rule On Elastix.....	5
Configure Inbound Rule On Elastix.....	6
Connect FEG4301 with Elastix	8
Create PCM Trunk Group on FEG4301.....	9
FEG4301 CALL Routing.....	10
Configure IP to PSTN on FEG4301	10
Configure PSTN to IP on FEG4301	11



Table of Figures

Figure 1: Create IVR on the Elastix	4
Figure 2: Create Peer SIP Trunk on the Elastix.....	5
Figure 3: Configure Outbound Rule on the Elastix	6
Figure 4: Configure Inbound Rule on Elastix.....	7
Figure 5: Connect FEG4301 with Elastix: Remote Address	8
Figure 6: Connect FEG4301 with Elastix: SIP Trunk Group	8
Figure 7: Create PCM Trunk Group on FEG4301	9
Figure 8: FEG4301 – IP to PSTN.....	10
Figure 9: IP to PSTN Settings	10
Figure 10: FEG4301 – PSTN to IP.....	11
Figure 11: PSTN to IP Settings	11



OVERVIEW

This document describes basic configuration to interconnect Elastix PBX 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.

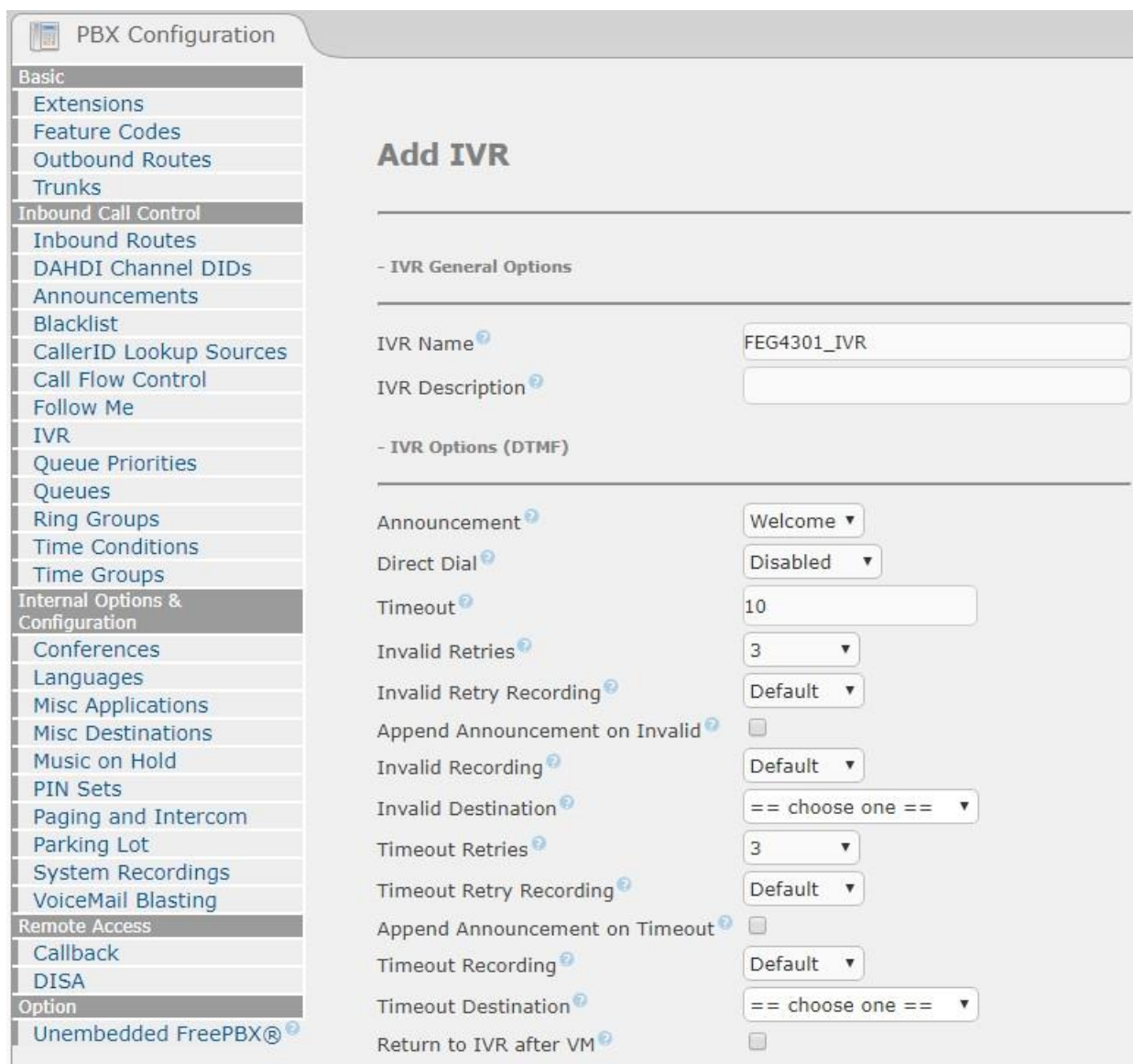


CONNECT Elastix TO FEG4301 USING PEER SIP TRUNK

Create IVR On Elastix

On the Elastix web GUI, create an IVR extension under **PBX->PBX Configuration->IVR**.

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



PBX Configuration

- Basic
 - Extensions
 - Feature Codes
 - Outbound Routes
 - Trunks
- Inbound Call Control
 - Inbound Routes
 - DAHDI Channel DIDs
 - Announcements
 - Blacklist
 - CallerID Lookup Sources
 - Call Flow Control
 - Follow Me
 - IVR
 - Queue Priorities
 - Queues
 - Ring Groups
 - Time Conditions
 - Time Groups
- Internal Options & Configuration
 - Conferences
 - Languages
 - Misc Applications
 - Misc Destinations
 - Music on Hold
 - PIN Sets
 - Paging and Intercom
 - Parking Lot
 - System Recordings
 - VoiceMail Blasting
- Remote Access
 - Callback
 - DISA
- Option
 - Unembedded FreePBX®

Add IVR

- IVR General Options

IVR Name [?] FEG4301_IVR

IVR Description [?]

- IVR Options (DTMF)

Announcement [?] Welcome ▾

Direct Dial [?] Disabled ▾

Timeout [?] 10

Invalid Retries [?] 3 ▾

Invalid Retry Recording [?] Default ▾

Append Announcement on Invalid [?]

Invalid Recording [?] Default ▾

Invalid Destination [?] == choose one == ▾

Timeout Retries [?] 3 ▾

Timeout Retry Recording [?] Default ▾

Append Announcement on Timeout [?]

Timeout Recording [?] Default ▾

Timeout Destination [?] == choose one == ▾

Return to IVR after VM [?]

Figure 1: Create IVR on the Elastix



Create Peer SIP TRUNK On Elastix

On the Elastix web GUI, create a peer SIP trunk under **PBX ->PBX Configuration ->Trunks**. In this example, the FEG4301 IP address is 192.168.99.237.

Add SIP Trunk

General Settings

Trunk Name [?]: FEG4301

Outbound CallerID [?]:

CID Options [?]: Allow Any CID ▾



Maximum Channels [?]:

Asterisk Trunk Dial Options [?]: Override

Continue if Busy [?]: Check to always try next trunk

Disable Trunk [?]: Disable

Dialed Number Manipulation Rules [?]

(prepend) + prefix | match pattern  

+ Add More Dial Pattern Fields Clear all Fields

Dial Rules Wizards [?]: (pick one) ▾

Outbound Dial Prefix [?]:

Outgoing Settings

Trunk Name [?]: FEG4301

PEER Details [?]:

```
host=192.168.99.237
type=peer
port=5060
context=from-trunk
insecure=very
qualify=yes
```

Figure 2: Create Peer SIP Trunk on the Elastix

Configure Outbound Rule on Elastix

On the Elastix web GUI, go to **PBX ->PBX Configuration ->Outbound Routes** to create a new outbound rule. This would allow the extension on the Elastix to reach numbers in ISDN network via the peer SIP trunk we just configured.



Add Route

Route Settings

Route Name [?]:

Route CID: [?] Override Extension [?]

Route Password: [?]

Route Type: [?] Emergency Intra-Company

Music On Hold? [?]

Time Group: [?]

Route Position [?]

Additional Settings

PIN Set [?]:

Call Recording [?]:

Dial Patterns that will use this Route [?]

(prepend) + 9 | [. / CallerID]

[+ Add More Dial Pattern Fields](#)

Dial patterns wizards [?]:

Trunk Sequence for Matched Routes [?]

0

1

2

Figure 3: Configure Outbound Rule on the Elastix

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 Elastix

On the Elastix web GUI, go to **PBX ->PBX Configuration ->Inbound Routes** 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.



Add Incoming Route

Add Incoming Route

Description [?]:

DID Number [?]:

CallerID Number [?]:

CID Priority Route [?]:

Options

Alert Info [?]:

CID name prefix [?]:

Music On Hold [?]:

Signal RINGING [?]:

Pause Before Answer [?]:

Privacy

Privacy Manager [?]:

Fax Detect

Detect Faxes [?]:

Language

Language [?]:

CID Lookup Source

Source [?]:

Call Recording

Call Recording [?]:

Set Destination

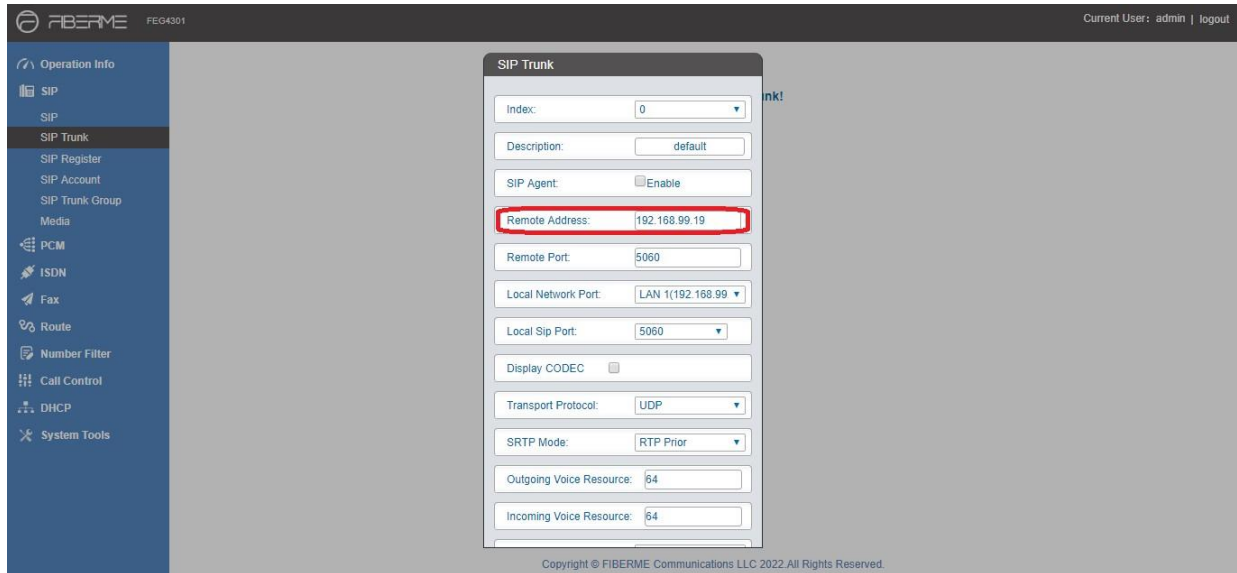
Figure 4: Configure Inbound Rule on Elastix

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



Connect FEG4301 with Elastix

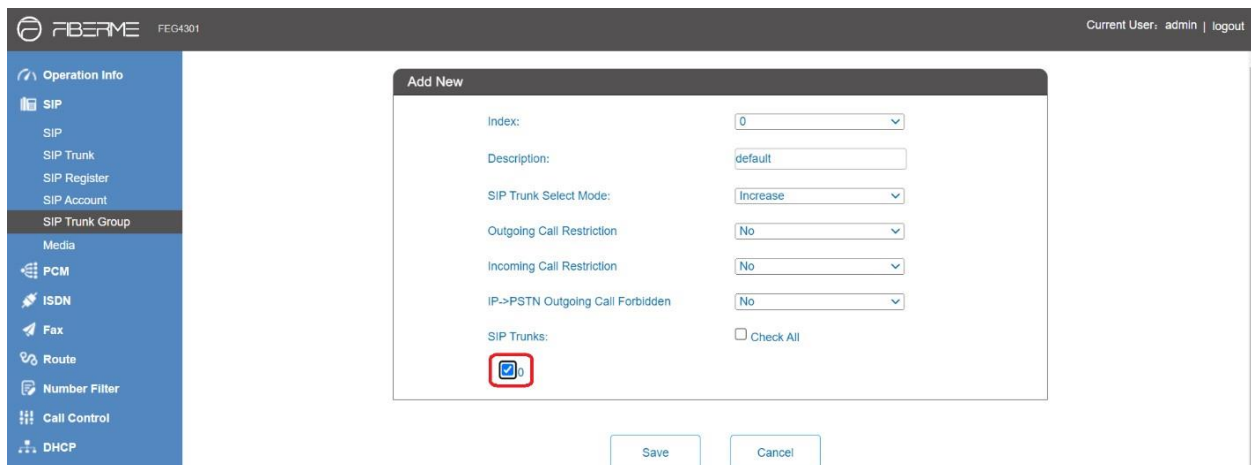
1. On the FEG4301 web GUI, go to the **VoIP->SIP Trunk** page and click on **Add New** button. Then, enter the IP address of the Elastix PBX that you are peering with in Remote Address. In the following example, Elastix has IP address 192.168.99.19.



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 rectangle and contains the value '192.168.99.19'. Other fields include Index (0), Description (default), SIP Agent (Enable), Remote Port (5060), Local Network Port (LAN 1(192.168.99)), Local Sip Port (5060), Display CODEC, Transport Protocol (UDP), SRTP Mode (RTP Prior), Outgoing Voice Resource (64), and Incoming Voice Resource (64). The footer contains the copyright notice: 'Copyright © FIBERME Communications LLC 2022.All Rights Reserved.'

Figure 5: Connect FEG4301 with Elastix: 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 'Add New' dialog box in the FEG4301 web GUI. The dialog 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 (Check All). The checkbox for index 0 is checked and highlighted with a red rectangle. At the bottom of the dialog are 'Save' and 'Cancel' buttons. The background shows the 'SIP Trunk Group' configuration page with the 'Add New' button highlighted.

Figure 6: Connect FEG4301 with Elastix: 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.

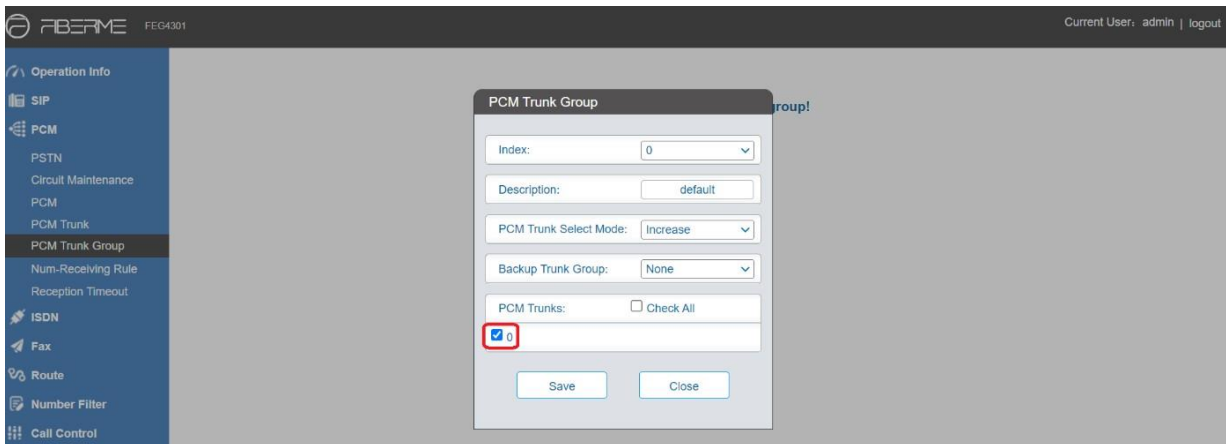


Figure 7: Create PCM Trunk Group on FEG4301



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”.

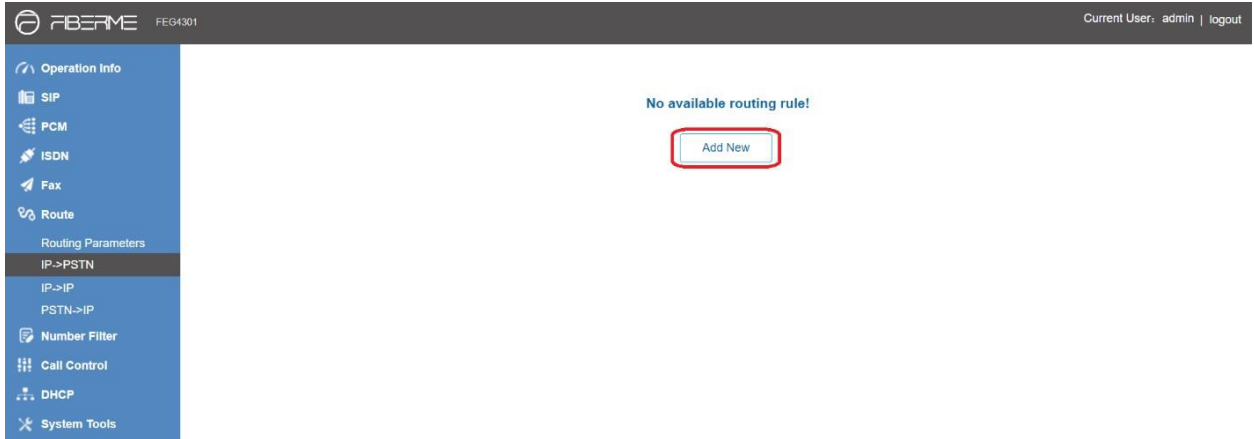


Figure 8: FEG4301 – IP to PSTN

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

The image shows a dialog box titled "IP->PSTN Routing Rule". It contains several input fields: "Index" with a dropdown set to "255"; "Call Source" with a dropdown set to "SIP Trunk Group [0]"; "CallerID Prefix" with a text input containing "*"; "CalleeID Prefix" with a text input containing "*"; "Call Destination" with a dropdown set to "PCM Trunk Group [0]"; "Number Filter" with a dropdown set to "none"; and "Description" with a text input containing "default". At the bottom are "Save" and "Close" buttons. Red boxes highlight the "Call Source" and "Call Destination" dropdowns.

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”.

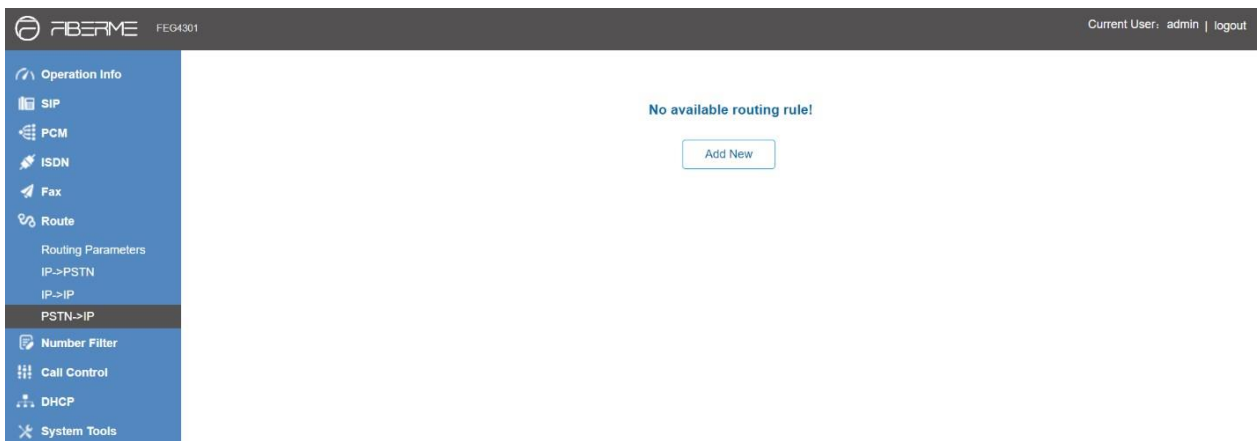


Figure 10: FEG4301 – PSTN to IP

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

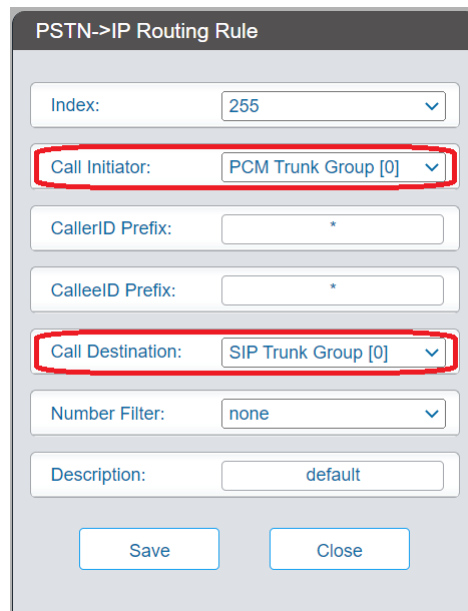
The image shows a 'PSTN->IP Routing Rule' configuration dialog box. It contains several fields: 'Index' (255), 'Call Initiator' (PCM Trunk Group [0]), 'CallerID Prefix' (*), 'CalleeID Prefix' (*), 'Call Destination' (SIP Trunk Group [0]), 'Number Filter' (none), and 'Description' (default). The 'Call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

Figure 11: PSTN to IP Settings

