

Connecting FXO Gateway to Elastix®

New Rock’s SIP based VoIP gateways (MX series gateways) with FXO ports are compatible with the open-source software Elastix® for unified communication solutions.

Scenario

The MX gateway with FXO ports and Elastix® are in the same LAN. A static IP address 192.168.250.99 is set for FXO gateway whose FXO ports connected to the PSTN lines and a static IP address 192.168.250.222 is set for Elastix®.

In this application note, two most common applications and their related configurations are described, including:

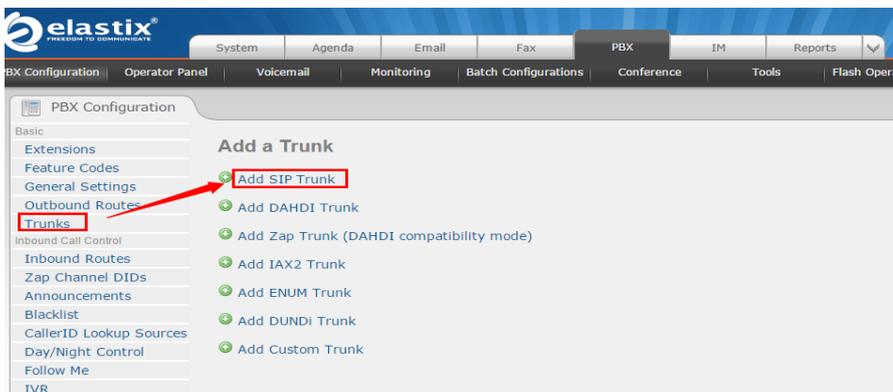
- Peer to Peer mode (without registration)
- Registration mode

After the following configurations (either peer to peer mode or registration mode), the gateway can route the calls from Elastix® to FXO ports for outbound, and can direct the incoming calls from PSTN to AA system of Elastix® via FXO ports, then an IVR prompt will be announced to PSTN, then callers on PSTN side can dial extension numbers after they heard the IVR prompt.

Peer to Peer Mode (Without Registration)

Configuration on Elastix®

1. On **PBX > PBX Configuration > Trunks** page, click **Add SIP Trunk** to add a SIP trunk



2. Fill in **Trunk Name** for both General and Outgoing Settings.

The screenshot shows the configuration interface for a trunk. The **General Settings** section includes fields for Trunk Name (highlighted with a red box), Outbound Caller ID, CID Options (set to 'Allow Any CID'), Maximum Channels, Disable Trunk (checkbox), and Monitor Trunk Failures (checkbox). The **Dialed Number Manipulation Rules** section has options for (prepend), prefix, and match pattern, along with buttons for adding more fields and clearing all fields. The **Outgoing Settings** section shows Trunk Name (highlighted with a red box) and PEER Details with a text area containing: `host=***provider ip address***
username=***userid***
secret=***password***
type=peer`

- 3. Fill in **PEER Details** (host = FXO gateway IP address; type=account type) on Outgoing Settings as follows:

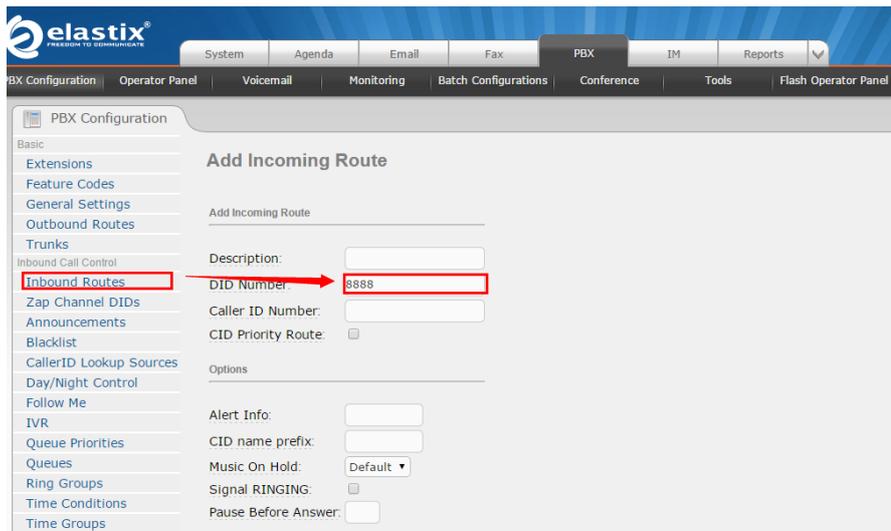
host=192.168.250.99

type=peer

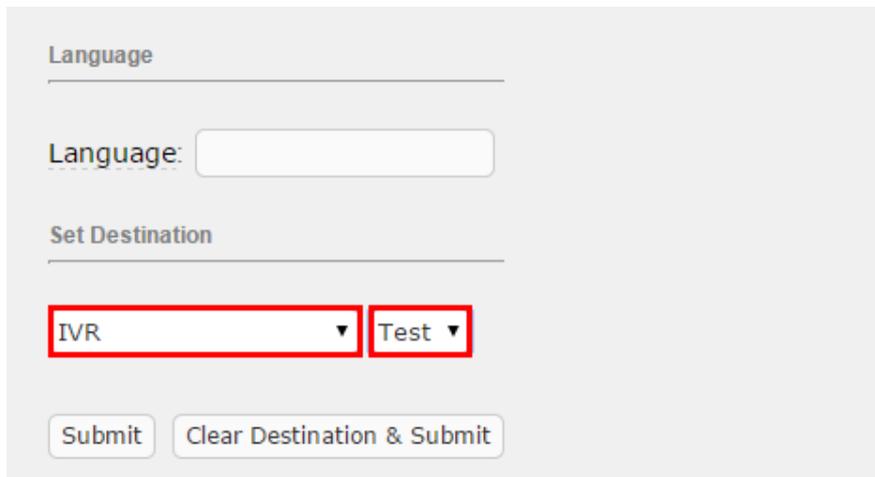
This screenshot shows the configuration page with the **PEER Details** text area filled in. The **Trunk Name** field is now 'Test_1'. The PEER Details text area contains: `host=192.168.250.99
type=peer` (the first line is highlighted with a red box). The **Outgoing Settings** section is visible, and the **Incoming Settings** section is partially visible at the bottom.

- 4. On **PBX > PBX Configuration > Inbound Routes** page, fill in **DID Number** (it can be any

number, just remember it, for you will use it when you configure the gateway).



5. Choose **IVR** in **Set Destination**.



Configuration on FXO Gateway

1. On **Basic>SIP** page, fill in Elastix® IP address in the Proxy server field.

The screenshot shows a configuration page with the following fields:

- Local signaling port: 5060 (Range: 1 - 9999, Default: 5060)
- Increments of port number: 5
- Registrar server: [Empty]
- Proxy server: 192.168.250.222 (e.g. 168.33.134.51:5000 or www.sipproxy.com:5000)
- Subdomain name: [Empty]
- Registrar mode: Per line
- User name: [Empty]
- Registrar password: [Empty]
- Registration expiration: 600 s

2. On **Routing > Routing Table** page, setup the routing rule for directing calls from Elastix® to PSTN via FXO ports in round robin order (Suppose FXO gateway has 8 FXO ports).

The screenshot shows the 'Routing table' configuration page with the following elements:

- Navigation tabs: Basic, Line, Trunk, **Routing**, Advanced, Call Status, Logs, Tools
- Sub-tabs: Digit map, **Routing table**, IP table
- Routing rule: IP X ROUTE 1-8/R
- Buttons: Save, Refresh

3. On **Basic > System** page, select **RFC 2833** in the **DTMF transmission method** field.

Basic	Line	Trunk	Routing	Advanced	Call Status	Logs	Tools
Status	Network	VLAN	System	SIP	MGCP	FoIP	
Off-hook timer	15	s (Range: 2 - 60, Default: 15)					
Interdigit timer	5	s (Range: 2 - 60, Default: 5)					
Complete entry timer	2	s (Range: 1 - 10, Default: 2)					
Codec	G.711U/20, G.711A/20	G.729A/20,G.711U/20,G.711A/20					
Hook-flash handle	Internal						
DTMF transmission method	RFC 2833						
RFC 2833 payload type	101	Range: 96 to 127, Default: 101, consistent with the opposite end (such as: softswitch platform)					
DTMF tone duration	100	ms (Range: 50 - 150, Default: 100)					
DTMF interdigit pause	100	ms (Range: 50 - 150, Default: 100)					
Min. DTMF detection duration	48	ms (The range must be 32 to 96 in multiples of 16)					
DTMF detection duration increment against talk-off	16	ms					

- On **Trunk >Feature** page, select **Binding** in the **Inbound handle** field, and fill in the corresponding DID number(the one you have remembered,it is 8888 here)on Elastix® server in the Binding numberfield.

Basic	Line	Trunk	Routing	Advanced	Call Status	Logs	Tools
Phone number	Feature	Batch	Advanced				
Trunk ID	FXO-3						
Phone number	8003						
Display as							
Registration	<input type="checkbox"/>						
Password							
Inbound handle	Binding						
Number	8888						
RFC6913	<input type="checkbox"/>						
<input type="checkbox"/> Polarity reversed signal detection	<input checked="" type="checkbox"/> Caller ID detection	<input type="checkbox"/> Outgoing call barring					
<input checked="" type="checkbox"/> Echo cancellation	<input type="checkbox"/> Connect signal delay	<input type="checkbox"/> Recording					
Save							

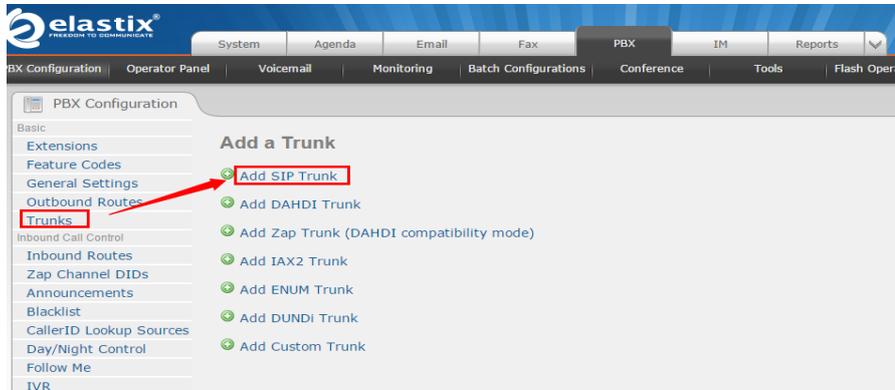
Dialing Method

- For making calls from PSTN to Elastix® through MX gateway, the caller dial the number of PSTN line connected to gateway’s FXO port, then dial the extension number of Elastix®after hearing the IVR.
- For making outgoing calls fromElastix® extensions, the extensions directly dial the outside number.

RegistrationMode

Configuration on Elastix®

1. On **PBX > PBX Configuration > Trunks**page, click **Add SIP Trunk** to add a SIP trunk



2. Fill in **Trunk Name** for both General and Outgoing settings.

A screenshot of the Elastix configuration page for a SIP Trunk. The page is divided into two main sections: 'General Settings' and 'Outgoing Settings'. In the 'General Settings' section, the 'Trunk Name' field is highlighted with a red box. Other fields include 'Outbound Caller ID', 'CID Options' (set to 'Allow Any CID'), 'Maximum Channels', 'Disable Trunk' (checkbox), and 'Monitor Trunk Failures' (checkbox). Below this is the 'Dialed Number Manipulation Rules' section, which includes a rule editor with 'prepend', 'prefix', and 'match pattern' options, and a 'Dial Rules Wizards' dropdown. The 'Outgoing Settings' section also has a 'Trunk Name' field highlighted with a red box, and a 'PEER Details' text area containing the following text:

```
host=***provider ip address***  
username=***userid***  
secret=***password***  
type=peer
```

3. Fill in **PEER Details** (host = FXO gateway IP address; username=SIP account; secret=password; type=account type) on Outgoing Settings as follows:

host=192.168.250.99

username=test

secret=abc123

type=friend

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: Test_1

PEER Details:

```
host=192.168.250.99
username=test
secret=abc123
type=friend
```

4. On **PBX > PBX Configuration > Inbound Routes** page, fill in **DID Number** (it can be any number, just remember it, for you will use it when you configure the gateway).

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PBX Configuration Operator Panel Voicemail Monitoring Batch Configurations Conference Tools Flash Operator Panel

PBX Configuration

Basic
Extensions
Feature Codes
General Settings
Outbound Routes
Trunks
Inbound Call Control
Inbound Routes
Zap Channel DIDs
Announcements
Blacklist
CallerID Lookup Sources
Day/Night Control
Follow Me
IVR
Queue Priorities
Queues
Ring Groups
Time Conditions
Time Groups

Add Incoming Route

Add Incoming Route

Description:

DID Number: 8888

Caller ID Number:

CID Priority Route:

Options

Alert Info:

CID name prefix:

Music On Hold: Default

Signal RINGING:

Pause Before Answer:

5. Choose **IVR** in **Set Destination**.

Language

Language:

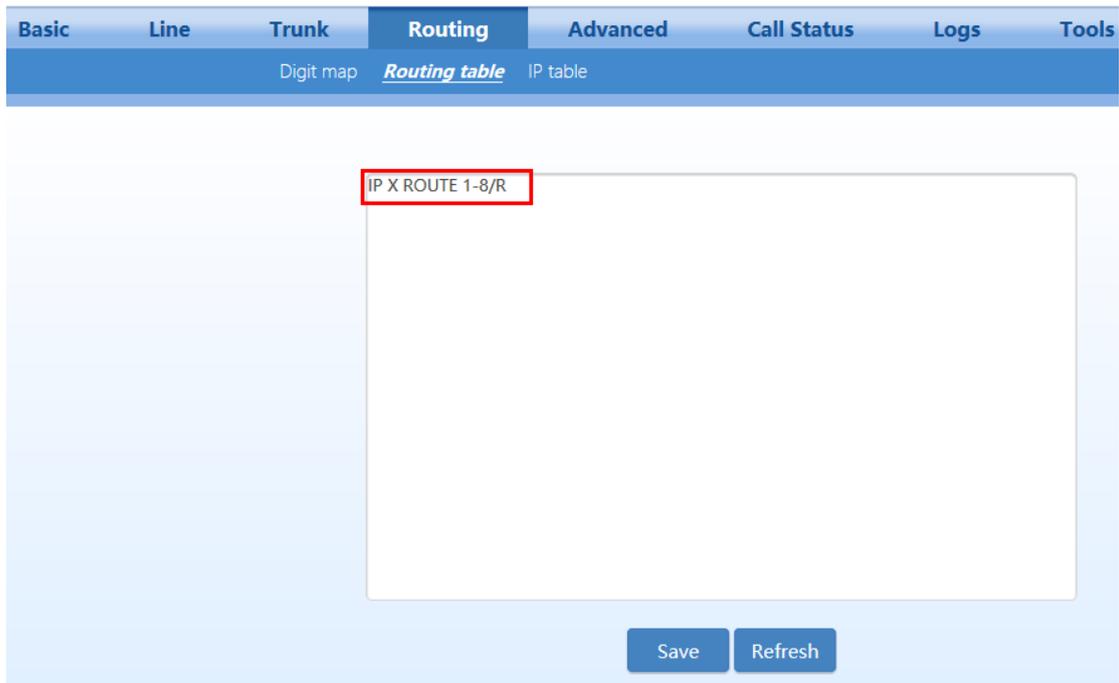
Set Destination

Configuration on FXO Gateway

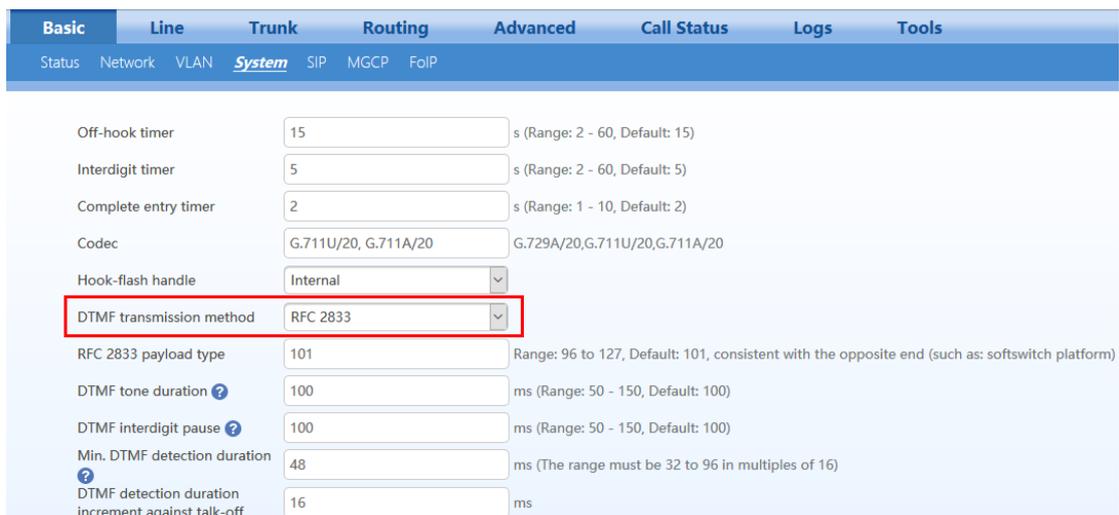
1. On **Basic >SIP**page, fill in Elastix®IP address in the Registrar and Proxy server fields,select **Per gateway**as the Registrar mode and fill in the User name and Registrar password fields (the same as the username and the secret in Elastix®).

Basic	Line	Trunk	Routing	Advanced	Call Status	Logs	Tools
Status	Network	VLAN	System	SIP	MGCP	FoIP	
Local signaling port	5060		(Range: 1 - 9999, Default: 5060)				
Increments of port number	5		?				
Registrar server	192.168.250.222						
Proxy server	192.168.250.222		e.g. 168.33.134.51:5000 or www.sipproxy.com:5000				
Subdomain name	<input type="text"/>						
Registrar mode	Per gateway		v				
User name	test						
Registrar password	••••••						
Registration expiration	600		s				

2. On **Routing > Routing Table** page, setup the routing rule for directing call from Elastix®to PSTN via FXO ports in round robin order(Suppose FXOgateway has 8FXO ports).



3. On **Basic > System** page, select **RFC 2833** in the **DTMF transmission method** field.



4. On **Trunk >Featurepage**, select **Binding** in the **Inbound handle** field, and fill in the corresponding DID number(the one you have remembered, it is 8888 here)on Elastix® server in the Binding numberfield.

Basic	Line	Trunk	Routing	Advanced	Call Status	Logs	Tools
Phone number	Feature	Batch	Advanced				
Trunk ID		FXO-3					
Phone number		8003					
Display as							
Registration		<input type="checkbox"/>					
Password							
Inbound handle		Binding					
Number		8888					
RFC6913		<input type="checkbox"/>					
<input type="checkbox"/> Polarity reversed signal detection		<input checked="" type="checkbox"/> Caller ID detection				<input type="checkbox"/> Outgoing call barring	
<input checked="" type="checkbox"/> Echo cancellation		<input type="checkbox"/> Connect signal delay				<input type="checkbox"/> Recording	
<input type="button" value="Save"/>							

Dialing Method

- For making calls from PSTN to Elastix® through MX gateway, the caller dial the number of PSTN line connected to gateway's FXO port, then dial the extension number of Elastix® after hearing the IVR.
- For making outgoing calls from Elastix® extensions, the extensions directly dial the outside number.