SIP Trunk Configuration for Broadvox

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Prerequisites

The Broadvox customer service provides the following communication parameters:

Parameter	Example	Explanation
BTN & Username:	`4801234560`	Typically, this is one of the DID numbers provided.
Password	`123456`	

DNS A records of SIP servers	`New York City, NY:`
	`nyc01-01.fs.broadvox.net`
	
	`Dallas, TX:`
	`dfw01-01.fs.broadvox.net`
	
	`Los Angeles, CA:`
	`lax01-01.fs.broadvox.net`
		These are the SIP servers on which the PBX must register.

DID numbers	`4801234560-4801234582`	These are the public telephony numbers allocated for the PBX by the provider.
Simultaneous calls	`10`	This is the maximum number of the simultaneous calls that can be made through three SIP trunks.

Configuration Process

STEP 1. Create Three SIP Trunks

It is necessary to create three SIP trunks. One for each Broadvox server. Each trunk may be created as it is described here.

Navigate to the Trunk menu entry in the PBX Settings and click the 'Add SIP Trunk' link. Define the trunk parameters as follows:

`Outbound Caller ID`	One of the received DID numbers can be placed here. In cases where the "Outbound CID" parameter is not defined, this DID number will be used as the Caller ID for the outbound calls from the PBX extensions. This parameter is optional.
`Maximum channels`	Indicate the number of simultaneous calls here.
`Dial Rules`	Broadvox requires the called numbers to be presented in the 10 digits format for North American calls and 011+international number for international calls. If the PBX users are used to dial 7 digits for local calls then it is possible to complete the number up to 10 digits here. For example: 702+XXXXXXX
`Trunk name`	We suggest you define it as `broadvox-ny,broadvox-la or broadvox-tx,` depending on the server address you will define in the host parameter (see below).

`PEER Details`	define it as follows:
	
	`host=dfw01-01.fs.broadvox.net`
	`context=from-trunk`
	`type=friend`
	`insecure=port,invite`
	`canreinvite=no`
	`qualify=no`
	`username=4801234560`
	`secret=123456`
	
	You should define the Broadvox server address located at the nearest to you place.
	
	Replace `4801234560` and `123456` with the user name and password that you received from Broadvox. If you agreed with Broadvox a static IP address of your PBX then it is possible to omit the `username` and `secret` parameters.
	
	If you would like to define a particular voice codec list then you should add the following two lines:
	
	`disallow=all`
	
	`allow=g729,ulaw,alaw`
	
	where, the `allow` parameter must contain the codec names prioritized in the desired order.

`USER Details`	Leave this field empty.

`Registration`	Define the parameters that will be used by Asterisk for SIP registration on the Broadvox SIP server. The registration string must be defined in the following format:
	
	`user:secret@domain/user`
	
	In our example the registration string would be:
	
	`4801234560:123456@dfw01-01.fs.broadvox.net/4801234560`

Finally, click the Submit button.

STEP 2. Define Outbound Route

Navigate to the `Outbound Routes` menu entry in the `PBX Settings` and click '`Add Route`'.

Define suitable `Dial Patterns` and select the 'SIP/`broadvox-nn`' trunks in the desired order in the '`Trunk Sequence`'. There are no special recommendation which trunk should have the highest priority but you can range the trunks according their geographical nearness to your place.

STEP 3. Define DID and CID

For the PBX extensions with DID numbers assigned, define the Direct DID and Outbound CID fields in the corresponding extension setting fields. If the Outbound CID field is empty, the number defined in the “Outbound Caller ID” supplied in the trunk configuration will be used as the caller ID for the outbound calls from this extension.

STEP 4. Apply Changes

Click the “`Apply Configuration Changes`” pink bar.

STEP 5. Verify Registration

Check that the PBX has been registered on the SIP server.

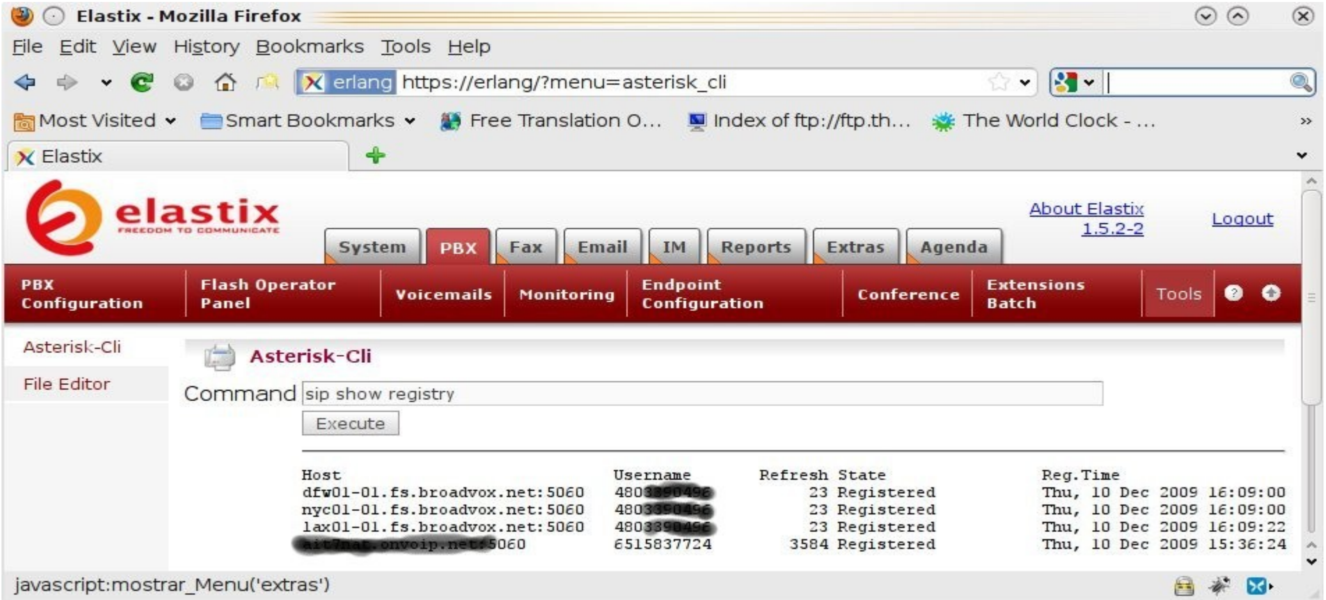
Connect to the Asterisk server via SSH and then connect to the Asterisk console by running the '`asterisk -r`' command. Check the output of the '`sip show registry`' command as follows:

```
MyPBX*CLI> sip show registry
Host                            Username     Refresh State         Reg.Time
dfw01-01.fs.broadvox.net:5060   4801234560   23      Registered    Thu, 10 Dec 2009 16:04:55

nyc01-01.fs.broadvox.net:5060   4801234560   23      Registered    Thu, 10 Dec 2009 16:04:55

lax01-01.fs.broadvox.net:5060   4803390496   23      Registered    Thu, 10 Dec 2009 16:04:52
```

Alternatively, it is possible to execute the command by using the “`Asterisk CLI`” option in the Elastix Web interface. Select the PBX tab, click Tools in the upper menu and then click “Asterisk-CLI” in the left side menu:



If the value for `State` is something other than '`Registered`' then check that the trunk parameters are defined correctly and your NAT/Firewall router doesn't block/distort the SIP messages. *Troubleshooting of SIP/NAT problems is not within the scope of this document.*