SIP Trunk Configuration for nexVortex

Document version: 1.0
Modification date: June 25, 2013

Prerequisites

The nexVortex customer service provides the following communication parameters:

Parameter	Example	Explanation
SIP server host name	`reg.nexvortex.com`	The SIP server which the PBX should use for registration. The customer will need to get this information from nexVortex.
Outbound Proxy server host name	`px3.nexvortex.com`	The Outbound proxy server which the PBX should use for outbound calls. The customer will need to get this information from nexVortex.
SIP-ID	`8234560430`	SIP user name that is used for registration and authentication on the SIP server. This is usually one of assigned DID numbers.
SIP-Password	`123456`	Case sensitive password
DID numbers	`6203101438-6203101447`	The public telephone numbers allocated for the PBX by the provider.

Configuration Process

STEP 1. Create a SIP trunk

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

Xorcom USA
145 S. Jefferson Ave., Suite G
Cookeville, TN 38501 USA
Tel: 1-866-967-2661
info.usa@xorcom.com

Business Telephony Solutions
www.xorcom.com

Xorcom Ltd.
Misgav Industrial Park, POB 60
D.N. Misgav 20174, Israel
Tel: +972-4-9951999
info@xorcom.com

`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`	Leave empty.
`Dial Rules`	nexVortex requires the national called numbers to be of 11 digits. A leading '1' is required. For example 18669672661.International numbers must be sent with 011 prefix. For example, 01133... for calling a number in France.
`Trunk name`	We suggest you define it as `nexVortex`, although any name may be used.
`PEER Details`	Define Trunk Name as nexVortex-out

```
type=friend
dtmfmode=auto
host=reg.nexvortex.com
username=8234560430
secret=123456
nat=yes
outboundproxy=px3.nexvortex.com
```

`USER Details` Define Trunk Name as nexVortex-in

```
type=peer
dtmfmode=auto
host=px3.nexvortex.com
username=8234560430
secret=123456
nat=yes
context=from-trunk
insecure=invite,port
```

`Registration`	Define the parameters that will be used by Asterisk for SIP registration on the nexVortex SIP server. The registration string should be defined in the following format:
	`peer?user:secret@host/extension`
	In our example the registration string would be:
	`nexVortex_out?`
	`8234560430:123456@reg.nexvortex.com/8234560430`

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/nexVortex' trunk in the '`Trunk Sequence`'.

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. Define the IP-PBX external IP address

The IP-PBX is behind a NAT router and should have a public static IP address assigned.
Whether Asterisk is talking to someone "inside" or "outside" of the NATted network.
This is configured by assigning the "localnet" parameter with a list of network addresses that are considered "inside" of the NATted network. Therefore, the following parameters should be defined in /etc/astersik/sip_general_custom.conf:

externip = <public IP address>

Multiple entries are allowed, e.g. a reasonable set is the following:

```
localnet=192.168.0.0/255.255.0.0          ; RFC1918 addresses
localnet=10.0.0.0/255.0.0.0                ; Also RFC1918
localnet=172.16.0.0/12                     ; Another RFC1918 with CIDR notation
localnet=169.254.0.0/255.255.0.0          ; Zero conf local network
```

That is for the sip trunk to send his sip messages back to the public IP address assigned to the IP-PBX.

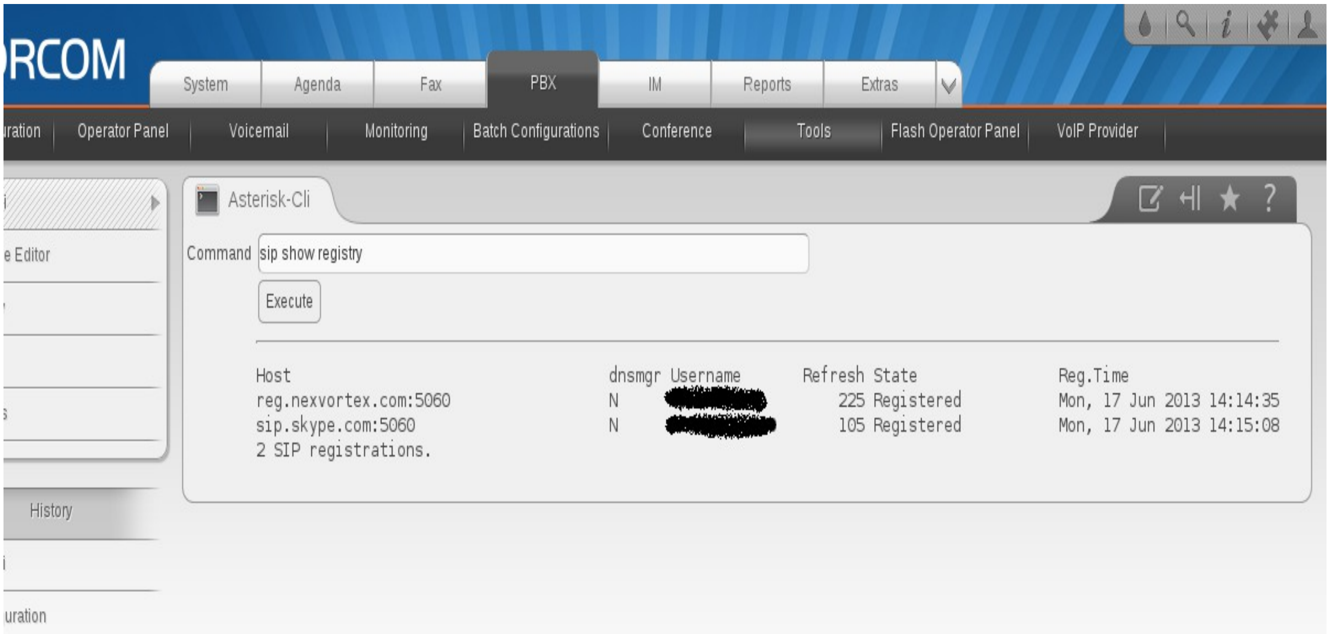
STEP 6. Verify Registration

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                     dnsmgr Username     Refresh  State      Reg.Time
reg.nexvortex.com:5060   N      8234560430   225      Registered Mon, 17 Jun 2013 14:10:50
```

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.*