nexVortex Setup Guide

Xorcom

October 2013

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Introduction

This document is intended only for nexVortex customers and resellers as an aid to setting up the Xorcom PBX software to connect to the nexVortex Business Grade SIP Trunking Service.

- Further Xorcom information can be found at http://www.xorcom.com/
- Further help may be obtained by emailing support@nexvortex.com.

If you find any errors in this document or have any suggestions, please email us at support@nexvortex.com so that we can make updates to this document.

Important! Your DNS Address
Your specific DNS address was provided in the Account Set Up email you received the day you opened your account. Your Authentication User ID and password are also in this email. If you need assistance locating this information, please contact support@nexvortex.com.

Note: For all instructions throughout this Guide, you must substitute your DNS address wherever xx.xx.xxx.xxx is referenced.

Proxy Servers
To connect to the nexVortex network, you will need to add our proxy address into your phone system or device. The address of our proxy server will be a fully qualified domain name (FQDN). It was automatically sent to you when your account was setup. If you no longer have this information or would like us to issue a new proxy key, please contact us at support@nexvortex.com.

Note: If your system does not support a fully qualified domain name format, please contact support for a list of valid IP addresses for your account.

Special Characters
Please note that special characters should not be used anywhere in SIP configurations. These include, but are not limited to, @#$%&! and spaces.

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Step 1: Trunk Configuration

Inbound service – You may receive SIP signaling from nexVortex from any of the following IP addresses:

- 66.23.129.253
- 66.23.138.162
- 66.23.190.100
- 66.23.190.200
- 209.193.79.80

Your PBX should be configured with as many trunks as necessary to field traffic from these IPs. If you need additional assistance ensuring your local PBX configuration meets this requirement, please contact technical support for your equipment directly.

Outbound service – The most efficient way to ensure redundancy for outbound calling is to utilize DNS SRV for routing traffic to nexVortex. At present, if your PBX supports DNS SRV, pointing to 'nexvortex.com' as your Proxy IP address is all that should be necessary to ensure outbound redundancy.

If your PBX does not support DNS SRV, hopefully it supports configuration of multiple outbound proxies. If so, you should configure px1.nexvortex.com or px3.nexvortex.com or as your primary proxy IP address, and px5.nexvortex.com as your secondary IP address. If you need additional assistance with DNS SRV or configuring multiple outbound proxy IPs on your PBX, please contact technical support for your equipment directly.

Step 2: Trunk Settings

Your trunks MUST be configured to present your provisioned E911 number(s) (The E911 settings you created for your account at nexVortex.com) for Emergency calls (911) or emergency TEST calls (311 or 933). Either a proper FROM or P-Asserted-Identity (preferred) header containing your provisioned Emergency number, if you require additional information, please contact support.

In order to provide the highest level of service availability possible, nexVortex utilizes an n+1 architectural model for our call processing. You will need to ensure that your network edge (router and/or firewall) is configured to accommodate this architecture.

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- 66.23.190.100
- 66.23.190.200
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You must ensure that each of these IPs is allowed to pass UDP 5060 traffic into your network and that this traffic is port-forwarded (if necessary) to the internal IP of your PBX.

You will also need to open the RTP or audio ports. This is different for each customer premise device. Please reference Xorcom for this detail. Your edge device must be configured to allow inbound RTP traffic on this port range from ALL IP addresses.

SIP Trunk Configuration for nexVortex

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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:

`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.nexvor tex.com username=823456 0430 secret=123456 nat=yes outboundproxy=px3.nexvor tex.com
USER Details	Define Trunk Name as nexVortex-in type=peer dtmfmode=auto host=px3.nexvor tex.com username=823456 0430 secret=123456 nat=yes context=from- trunk insecure=invi te,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 Inbound Routes

The nexVortex servers will send calls to the PBX with a DID allocated on it (ref. “Prerequisites”, “DID Numbers”). Therefore, it is necessary to configure one or more Inbound routes for those DIDs. For the PBX extensions with DID numbers assigned, define the Direct DID and Outbound CID fields in the corresponding extension setting fields. If a Direct DID is defined then an inbound route will be created automatically for this number. 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.

For other DIDs you can define one or more inbound routes in the PBX Configuration->Inbound Routes dialog.

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.

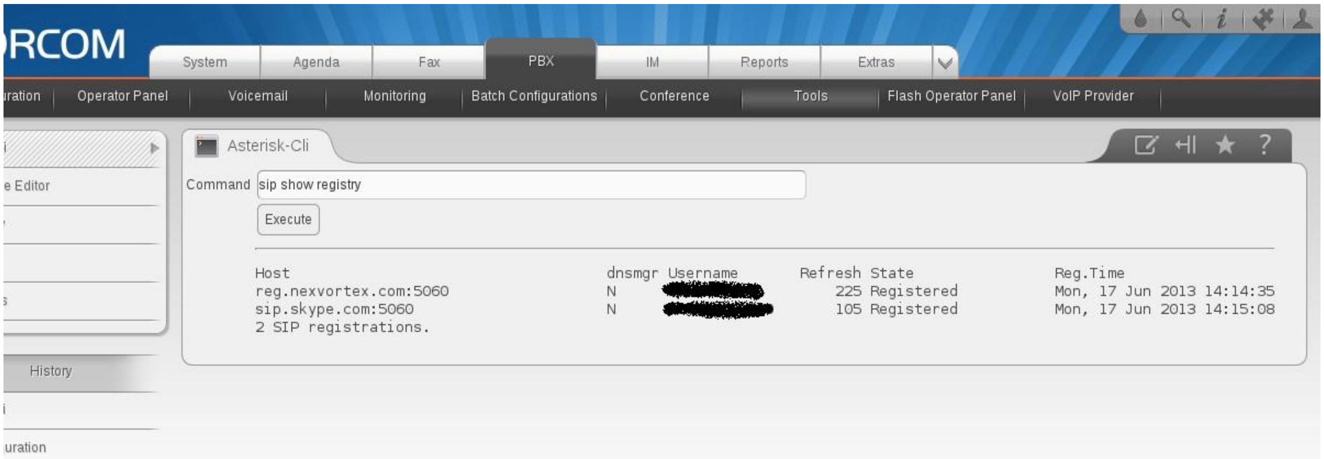
STEP 6. Verify Registration

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*.

Troubleshooting

Customer System will not register with nexVortex:

- Check the system is pointing at xx.xx.xxx.xxx
- Check UDP port 5060 is open on the firewall
- Check NAT translation is correct between LAN private IP address and public IP address
- Check you have the correct proxy user name and password configured.

Customer System cannot make a call:

- Check the system is pointing at xx.xx.xxx.xxx
- Check UDP port 5060 is open on the firewall
- Check NAT translation is correct between LAN private IP address and public IP address
- Check you have the correct proxy user name and password configured.

Customer System cannot receive a call:

- Some systems require our IP Address xx.xx.xxx.xxx for verification to be configured
- Check UDP port 5060 is open on the firewall
- Check NAT translation is correct between LAN private IP address and public IP address
- Check that you have setup the IP route for the number correctly with nexVortex. This is done through the customer or reseller Partner Connect portal->Settings-> Number Routing
- Check that the dial plan is configured to route the number to a valid location on the customer system.

One way audio or no audio after call is setup:

- Check the RTP audio ports are open on the firewall.

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