General Guidelines for SIP Trunking Installations

1) ***How do I setup my SIP trunk for inbound/outbound calling?***
We authenticate IP-PBX SIP Trunking traffic by:
IP Authentication (IP address) ***or***
Digest Authentication (account and SIP password)

Establish a SIP trunk with our US proxy server **169.132.196.33** and input your IP address into our portal or register your switch with us. Alternatively, if your switch is not in Central or North America, you can use one of our international POPs to reduce transit delays:

For our UK POP point to **213.166.103.6**

For our HK POP point to **111.235.152.141**

For our BR POP point to **206.20.196.31**

For our PE POP point to **206.20.196.33**

For our AR POP point to **206.20.196.34**

IP Authentication *(IP Address)*
The IP Authentication method is normally simpler to provision and should be used only when you have a static IP Address. It is also somewhat more secure since your SIP trunk can only be used from the IP Address you provide.

If you are using a web-based Asterisk PBX (like FreePBX), IP Authentication setup is slightly different:
In "Outgoing Settings", name the section "out-1"
Then, in "Peer Detail", enter the following:
type=peer
port=5060
nat=auto
insecure=invite
ignoresdpversion=yes
host= **169.132.196.33**
dtmfmode=rfc2833
context=from-trunk
canreinvite=no
allow=ulaw
allow=alaw
allow=g729
In "Incoming Settings", name the section "in-1" in "User Context". Then, in "User Detail, enter the following:

disallow=all
type=peer
port=5060
nat=auto
insecure=invite
host=**169.132.196.33**
dtmfmode=rfc2833
context=from-trunk
canreinvite=no
allow=ulaw
allow=alaw
allow=g729

Digest Authentication *(account & SIP Password)*
On open source applications (such as Asterisk), you can setup your SIP trunk for digest authentication as follows:
Peer Detail
username= <account>
type=peer
secret= <sip password>
progressinband=never
port=5060
nat=auto
insecure=very
ignoresdpversion=yes
host= **169.132.196.33**
dtmfmode=rfc2833
context=from-trunk
canreinvite=no
allow=g729&g711

User Detail
username=<account>
user=<>
type=user
port=5060
context=from-pstn
canreinvite=no
allow=g729&ulaw&alaw

Register String: register=>account:password@**169.132.196.33**

*For utilizing our UK POP, anywhere you see **169.132.196.33**, replace with **213.166.103.6** & for utilizing our Hong Kong POP replace with **111.235.152.141.** There are several GUI interfaces for Asterisk that simplify the installation process.

These interfaces allow administrators to view, edit and change most configurations via a Web interface. Unless you are an advanced Asterisk user, we highly recommend downloading one of the following GUI interfaces:

Elastix
http://www.elastix.org/
PBX in a Flash
http://pbxinaflash.net/
AsteriskNOW
http://www.asterisk.org/asterisknow/

2) ***Which CODECs do you support?***
We support G711and G729. Generally speaking, we recommend that our customers offer G711 as well as G729 in their initial SIP INVITE to us.

3) ***How do I setup my dialing plan for outbound calling?***
You can choose a dialing plan when provisioning the account through our Partner Resource Center Web site. If you choose the "Universal" dial plan in the Partner Resource Center, you can then dial in the following formats:
Country Code+ Phone Number
011+Country Code+ Phone Number
00+ Country Code + Phone Number.

US dialing plans are setup in the North American Numbering Plan (NANP) format of 1 + area code + 7 digit number (for US calls) and 011 + country code + number (for non-US calls). The 1 prefix should be used on all US calls. The 011 prefix should be used on all non-US calls.

For more information on the NANP, please visit http://www.nanpa.com/index.html
If you choose to setup a non-US dialing plan, you will first have to configure your service account on our Partner Resource Center Web site with the specific dialing plan of your choice. Then you would dial exactly the same way you typically dial from within the country, with local calls being dialed in the way you normally dial local calls in country.

4) ***Which format should I use when setting up Caller ID in my switch?***
We recommend using the PAID (P-Asserted-Identity) option for CLI as per RFC 3325. Our platform also supports RPID (Remote-Party-ID). For more information, please visit
http://www.ietf.org/rfc/rfc3325.txt

5) ***What ports should I open in my router/firewall?***
For best results please allow traffic from: - 169.132.196.33 and 206.20.196 for US - 213.166.103.6 for UK - 111.235.152.141 for Hong Kong
In addition, please allow all RTP traffic from any IP Address ports 20000-24000 UDP.

6) ***In what format should I configure the phone number for inbound routes?***
In order to receive inbound calls, you will have to build an inbound route on your switch and map it to a valid extension or ring group. Please make sure that you have configured the inbound route properly on your switch. Also, please make sure that you are allowing the inbound SIP traffic to pass to your switch from our proxy server.

We will send all inbound calls in the North American Numbering Plan (NANP) format of 1 + area code + 7 digit number (US calls) and 011 + country code + number (non US calls). This is how virtual phone numbers (DIDs) will display in the Partner Resource Center. You should input the DID in your inbound route in the exact format displayed in the panel.

7) ***Do you offer location based proxies so I can register with a proxy closer to my location?***
We currently offer four proxies for registration that will send both SIP and media packets from that location. However, registering with the default **169.132.196.33** proxy will also route media traffic depending on your location.