Subject:	SIP Trunking turnups – General guidelines	Date:	December 12th, 2011
	Version 1.0 – Asterisk/Open Source Guide		

Below is a list of general guidelines for new SIP Trunking turnups that our customers + internal IDT staff should follow.

SIP Trunking turnups – General guidelines

1. How do I setup my SIP trunk for inbound / outbound calling?

On our IDT Platform, we authenticate the IPPBX Sip trunking traffic by service account and pin. After you decide which switch platform to use, you will need to establish a SIP Trunk with our proxy server ippbx.net2phone.com port 5060 and register your switch with us.

On open source applications such as Asterisk, you could setup your SIP trunk as follows.

Peer detail

username= <account>
type=peer
secret= <pin>
progressinband=never
port=5060
nat=auto
insecure=very
ignoresdpversion=yes
host=ippbx.net2phone.com
dtmfmode=rfc2833
context=from-trunk
canreinvite=no
allow=g729&g711&g723

user detail

username=<account>

```
user=<>
type=user
port=5060
context=from-pstn
canreinvite=no
allow=g729&g711&g723
```

Register string

acct:pin@ippbx.net2phone.com/siptrunking

If you are not an advanced user of Asterisk, we highly recommend the use of one of below GUI interfaces for SIP Configuration

There are several GUI interfaces for Asterisk that simplifies installation of Asterisk. These interfaces allow administrators to view, edit and change most aspects of Asterisk via a web interface.

You can download free GUI versions of Asterisk from one of the below links.

Download Elastix
http://www.elastix.org/

Download PBX in a Flash
http://pbxinaflash.net/

Download AsteriskNOW
http://www.asterisk.org/asterisknow/

On other types of switch platforms, we recommend that you configure your switch as per your vendor guidelines (see appendix A below). Please note that IDT cannot provide direct technical support for customer SIP Trunking switch platforms. If your switch is not working as expected, you may need to contact the device manufacturer for technical support

2. What codecs does IDT support?

IDT supports G711, G729 as well as G723 codecs. Generally speaking, we recommend that our customers offer G711 as well as G729 codec in their initial SIP INVITE to us.

3. How do I setup my dial plan for outbound calling?

For the SIP Trunking service, you have the option to choose dial plan when provisioning the account on our Partner Resource Center website. In most cases, the dial plan is setup as a US dial plan which means that you would have to setup your dial plan as follows.

- NANP format (1+10 dig and 011+CC+..)

The 1 prefix should be used on all US calls. The 011 prefix should be used on all non US calls.

If you choose to setup your dial plan using an in country dial plan, you can do so but you will first have to configure your service account on our Partner Resource Center website with the specific dial plan that you prefer to use.

NANP stands for North American Numbering Plan.

http://www.nanpa.com/index.html

4. What format should I use when setting up Caller ID in my switch?

We prefer that our customers use PAID (P-Asserted-Identity) option for CLI as per RFC 3325. Our platform also supports RPID (Remote-Party-ID) as well.

http://www.ietf.org/rfc/rfc3325.txt

5. What DTMF settings should I set on my switch

We recommend that you enable RFC 2833 payload type 101 for DTMF. Please note that DTMF is supported over our Platinum routes (prefix 99901) only.

6. How can I capture a SIP trace on my switch?

When setting up a new SIP trunk with a provider or troubleshooting call failures it's important to be able to capture a signaling trace of an outbound call. This is also important when troubleshooting SIP registration issues with a new provider.

In Asterisk for example, you can turn on SIP debugging via the Asterisk CLI using the sip set debug commands for example.

Sip set debug peer on   - turns on SIP debugging globally showing all SIP traffic to and from the Asterisk gateway

Sip set debug IP xxx.xxx.xxx.xxx - allows you to debug only message to and from a particular IP address

Sip set debug off – Turns off all SIP debugging.

Please be familiar with how to turn on debugging on your specific switch.

If you don't want to enable debugging on your switch, you can use a network protocol analyzer such as Wireshark to capture the SIP and media traffic on your calls. To learn more about Wireshark, please visit http://www.wireshark.org/ This site includes step by step videos on how to setup wireshark on your network.

7. For fax transmissions, which protocol is supported by IDT.

For fax transmissions, generally speaking we support T.38 protocol as well as G711 pass-through. The majority of our carriers prefer to negotiate using T.38 protocol.

8. I am not able to receive inbound calls after setting up a SIP trunk with IDT. What should I do?

In order to receive inbound calls, you will have to build an inbound route on your switch + 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 ippbx.net2phone.com.

Appendix A

Support links to various switch vendor platforms

Asterisk – Support Forum

http://forums.asterisk.org/

3CX   - Support Forum

http://www.3cx.com/forums/

Avaya – Support Forum

http://support.avaya.com/css/appmanager/public/support

Elastix - Support Forum

http://www.elastix.org/index.php/en/component/kunena/

Trixbox – Support Forum

http://fonality.com/trixbox/forum

Grandstream - Support Forum

http://forums.grandstream.com/forums/

Cisco - Support Forums

http://www.cisco.com/cisco/web/support/index.html

FreePBX - Support Forum

http://www.freepbx.org/forums

Talkswitch - Support Forum

http://www.talkswitch.com/us/en/support/

AudioCodes – Support

http://www.audiocodes.com/support

Voipswitch – Support

http://www.voipswitch.com/

IPsmarx - Support

http://www.ipsmarx.com/Support.html

Sonus Networks - Support

http://sonusnetworks.force.com/PortalLoginPage

PortaOne – Support

http://portaone.com/support/portacare/