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# Skype connect and Asterisk

## General Configuration Guide Skype for SIP and Asterisk

If you are new to SIP, Asterisk is a useful, open-source (GPL) platform with which to test and experiment with the Skype for SIP. This is a guide on how to install Skype for SIP on a system agnostic or "vanilla" Asterisk server.

To install Asterisk on your server, please see the Digium documentation here <http://www.asterisk.org>.

This configuration guide is based on Debian Linux (Lenny 64bit). With a basic installation of Debian you can install Asterisk by issuing the following APT command at the command line:-  
apt-get  
install asterisk

## Configuration Files for Vanilla Asterisk

In configuring Skype for SIP on a vanilla Asterisk system we are primarily concerned with two configuration files:-

-  
sip.conf  
(located in the /etc/asterisk/ directory)The  
sip.conf file holds the registration details for the Skype for SIP  
channel

-  
extensions.conf  
(located in the /etc/asterisk/ directory)The extensions.conf  
holds the dial plan telling Asterisk what to do with incoming and  
outgoing calls.-

Let's  
do a walkthrough of the configuration steps.

## Configuring the sip.conf File

### Step 1

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The sip.conf file has two sections that need to be completed. The "General" section (denoted in the file with the [general] heading) and peer section denoted in the file with the [peers] heading.

In the General section we need to add a "register" line. This tells Asterisk to register with Skype at the Skype local point of presence.

Add the following, under the "[general]" section in the file, substituting your 9905xxxx number and password with your actual credentials for the Skype for SIP profile you wish to use. Your SIP Profile details can be found in the Skype Business Control Panel (BCP):-

```
register
=> 99051000xxxxxx:PaSsW0rD@sip.skype.com/99051000xxxxxx
```

Step

2

To

ensure that we also receive the callerID from Skype clients we also should add:-

```
trustpid
= no
sendrid
= yes
```

Step

3

Next,

we add a section for the peer, in the "[peers]" section of the sip.conf file. Again we substitute the 9905xxxxx number and password with the SIP Profile credentials from the Skype Business Control Panel (BCP):-

```
[99051000xxxxxx]
type
    = peer
username
    = 99051000xxxxxx
fromdomain
    = sip.skype.com
fromuser
    = 99051000xxxxxx
realm
    = sip.skype.com
host
    = sip.skype.com
dtmfmode
    = rfc2833
secret
    = PaSsW0rD
nat
    = no ;This
should be set to reflect your network NAT configuration
canreinvite
    = no
insecure
    = invite
qualify
```

```
= yes
disallow
= all
allow
= alaw
allow
= ulaw
;allow
= g729 ; Uncomment
this if you have G729 licences
amaflags
= default
trustpid
= no
sendrpid
= yes
context
= skype_in
```

Please  
Note:

If  
your Asterisk PBX is behind a NAT device, you should set `&ldquo;nat = yes&rdquo;` in this section.

If  
your Asterisk PBX has a dedicated internet IP address, set this to  
`&ldquo;nat = no&rdquo;`.

Step  
4

After  
setting these changes, reload the Asterisk's SIP module by typing:-

```
asterisk
-rx "reload chan_sip.so"
```

`&hellip;&hellip;`.at  
the command line.

Step  
5

After  
the SIP Module has reloaded enter asterisk  
-rx "sip show peers" at the command line, which  
should return:

```
pbx*CLI>
sip show peers
Name/username
      Host Dyn Nat ACL Port    Status
99051000xxxxxx/99051000xx
  193.120.218.68      5060    OK (52 ms)
```

Then  
enter asterisk  
-rx `&ldquo;sip  
show registry&rdquo;`  
which should return:

```
pbx*CLI>
sip show registry
Host
      Username    Refresh State
Reg.Time
```

---

```
sip.skype.com:5060
    99051000xxxx 105 Registered day, dd mmm
yyyy hh:mm:ss
```

If  
you see output similar to the above, then you are registered to the  
Skype SIP gateway and ready to make and receive calls.

We  
now need to setup the extensions.conf so that we have a dialplan  
setup and Asterisk knows how to deal with incoming and outgoing  
calls.

Configuring  
the extensions.conf File

The  
extensions.conf file requires a "context" and an "extension"  
to be added for incoming Skype calls, plus an extension to be added  
to the context  
that users use for outgoing calls.

Incoming  
"context";

Add  
the following lines to the [context] section of extensions.conf,  
substituting 9905xxxxxxx with the 9905 number for the SIP Profile.  
Again you can find the details of your Skype SIP Profiles in the  
Skype BCP:-

```
[skype_in]
exten
=> 99051xxxxxxx,Noop(${CALLERID(name)} , ${CALLERID(num)})
exten
=> 99051xxxxxxx,n,Dial(SIP/100,30,t,r)
exten
=> 99051xxxxxxx,n,voicemail(100|u)
```

This  
is a simple "vanilla" context that shows us the callerID name and  
number, dials extension 100 for 30 seconds and finally, if  
unanswered, goes to voicemail. This sequence will need to be amended  
to suit your requirements. If you are planning on having many SIP  
Profiles or Online Numbers that all need to end up at the same  
destination, or the destination is decided by the Skype Business  
Account that the online number is registered against, a more  
complicated Dialplan can be used. For example:-

```
[skype_in]
exten
=> 99051xxxxxxx,1,Noop(${CALLERID(name)} , ${CALLERID(num)})
exten
=> 99051xxxxxxx,n,Queue(sfs|r||40)
exten
=> 99051xxxxxxx,n,voicemail(100|u)
```

Outgoing  
"Context";

The  
outgoing context must be included in the context for your user's  
phones. Usual security measures apply. Do not include this in a  
context for incoming calls.

---

[skype\_out]

```
exten
=> _90Z.,1,Set(CALLERID(num)= 99051xxxxxxx)
exten
=> _90Z.,n,Dial(SIP/0044${EXTEN:2}@99051xxxxxxx)
```

```
exten
=> _900.,1,Set(CALLERID(num)= 99051xxxxxxx)
exten
=> _900.,n,Dial(SIP/${EXTEN:1}@99051xxxxxxx)
```

In  
the sip.conf add the following to create user 100

```
[100]
secret=secret
mailbox=100
callerid="myskypetrunk"
<100>
type=friend
host=dynamic
context=international
;nat=no
nat=yes
canreinvite=no
dtmfmode=rfc2833
pickupgroup=1
callgroup=1
subscribecontext=default
notifyringing=yes
disallow=all
;allow=alaw
allow=ulaw
allow=gsm
```

in the extensionsn.conf add the following to the default context

```
exten
=> _XXX,1,Dial(SIP/${EXTEN},20)
```

Also  
create a context called international

```
[international]
include
=> default
include
=> skype_out
```

This  
will allow phones with the international context to make outgoing calls and call other users.