How to use SIP trunking to connect a PBX to an extension

Applies to VoipNow 3 and higher!

This article describes the SIP trunking feature and how to use it to connect a PBX to an extension. Also, it includes a set of recommendations with examples for seiting up Asterisk to act as a PBX.

Enable SIP Trunking for a VoipNow extension

When enabled at extension level, this VoipNow feature allows you to connect a PBX to the extension (when a DID assigned to that extension is called, it is passed further to the PBX in the **SIP:To** header). This option was added because some customers wanted to connect different PBX systems to VoipNow.

In order to enable this feature, you need a SIP trunking license. If the license is valid, in the **Extension Provisioning and SIP** page, under the **SIP Preferences** area, you'll find the **PBX is connected to this extension [] Enable for SIP trunking service** option. Select this checkbox, and you will be able to connect a different PBX to that extension.

How to configure Asterisk to act as a PBX

For this, you will need Asterisk box. There are plenty of installation tutorials on the web, so you shouldn't find any difficulty in installing Asterisk.

In our example, `./configure --prefix=/opt` is used to set `/opt/` as a destination directory for the Asterisk files.

After completing the installation process, you will need to edit the files below:

Example of `/opt/etc/asterisk/modules.conf`

```
[modules]
autoload=yes
noload => pbx_gtkconsole.so
noload => pbx_spool.so
noload => chan_skinny.so
noload => res_smdi.so
noload => pbx_ael.so
noload => app_voicemail.so
load => res_musiconhold.so
noload => chan_alsa.so
noload => chan_console.so
```

Example of `/opt/etc/asterisk/sip.conf`

```
[general]
defaultexpirey=3600
maxexpirey=3600
disallow = all
allow = g729
allow = ulaw
allow = alaw
bindaddr = asterisk_server_ip
port = 5060
context = phones
nat = no
domain = your_server_ip
register => 0003*001:password@voipnow_server_ip/0003*001

[mysip]
fromuser = 0003*001
fromdomain = asterisk_server_ip
defaultuser = 0003*001
authuser = 0003*001
dtmfmode = rfc2833
dtmf = rfc2833
disallow = all
allow = g729
allow = ulaw
allow = alaw
type=peer
host = voipnow_server_ip
qualify = yes
nat = no
context = from-voip-provider
canreinvite = yes

[5000]
type=friend
defaultuser = 5000
secret = secret
qualify=yes
nat=no
host=dynamic
canreinvite=no
disallow=all
allow=alaw
context=phones
```

The example above registers the local Asterisk PBX to the VoipNow system using extension `0003*001`. Make sure the extension has SIP trunking enabled on the VoipNow server. `5000` is a local Asterisk extension that will be used for both incoming and outgoing calls.

Example of `/opt/etc/asterisk/extensions.conf`

```
[general]
static=yes
writeprotect=no
clearglobalvars=no

[phones]
exten => 5000,1,Dial(SIP/5000)
exten => _X.,1,Dial(SIP/${EXTEN}@mysip)
exten => _X.,2,Hangup

[from-voip-provider]
exten => 18002304043,1,Dial(SIP/5000)
exten => _X.,1,Congestion()
```

This is a basic example of Asterisk dialplan that can place outgoing calls and receive incoming calls. `5000` is the only local extension. The rest of the prefixes matching `_X.` get forwarded through the `mysip` channel. The`[from-voip-provider]` context is used for incoming calls.

To launch Asterisk, run the following:

```
/opt/sbin/asterisk -f -g -U asterisk -G asterisk
```

Test the configuration

For example, if you want to register the `5000` extension using a X-Lite softphone, you need to open its **SIP accounts Properties** menu page and set:

- **User name**: 5000
- **Password**: secret
- **Authorization user name**: 5000
- **Domain**: asterisk_server_ip

To call a different extension (e.g. `0003*002`) from the Asterisk PBX, you need to simply dial `0003*002`).

To take incoming calls via VoipNow on extension `5000`, the `[from-voip-provider]` context needs to be added. When configuring your `/opt/etc/asterisk/extensions.conf` file, you must replace the `18002304043` example with the DID number assigned to your account by your SIP provider.

Note that `18002304043` is a DID number assigned in the VoipNow interface for the specific extension with enabled SIP trunking.

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