# ASTERISK

optimum.

**SIP Trunking using Optimum Business**
**SIP Trunk Adaptor and the Asterisk IP PBX**
**Version 1.2.10**

## Goal

The purpose of this configuration guide is to describe the steps needed to configure the Asterisk IP PBX for proper operation with Optimum Business SIP Trunking.

## Prerequisites

Follow the instructions in the Optimum Business SIP Trunk Set-Up Guide left by the Optimum Business technician at installation. If you do not have the Set-Up Guide you can download it at www.optimumbusiness.com/SIP.

## Asterisk IP PBX Configuration

The steps below describe the minimum configuration required to enable the Asterisk IP PBX to use Optimum Business SIP Trunking for inbound and outbound calling. Please refer to the Asterisk product documentation for more information on other advanced PBX features. This configuration is based on Asterisk version 1.2.10. Note: Downloading and compiling Asterisk source code is beyond the scope of this document.

This configuration guide will provide two sample Asterisk configuration files: sip.conf and extensions.conf. These files are used to configure the station side SIP phones and the SIP trunk interface that connects to the Optimum Business SIP Trunk Adaptor.

This configuration guide assumes that the Asterisk PBX uses an IP address of 192.168.1.166 with station side SIP phones using DID phone numbers of 402-312-3982 to 402-312-3985. The pilot SIP Trunk DID in this example is 402-312-3982 and the Optimum Business SIP Trunk Adaptor's LAN interface uses an IP address of 192.168.1.1.

## Asterisk Configuration File: sip.conf

This configuration file is used to configure the Asterisk SIP trunk interface. The file below illustrates how to configure the Asterisk for either registration mode or static/non-registration mode of operation. The "**register**" configuration line that follows the **boldface** type below requires the pilot DID. Registration mode configures the Asterisk PBX to register the SIP trunk with the locally attached Optimum Business SIP Trunk Adaptor. The "**dtmfmode**" line under the "general" section must be set to **auto** in order for the auto-attendant to understand both in-band and out-of-band DTMF.

```
; sip.conf
;----------
[general]
dtmfmode=auto
context=default
srvlookup=yes
port=5060
bindaddr=192.168.1.166
maxexpiry=660
```

**;To configure the PBX to register with the Optimum Business SIP Trunk Adaptor, uncomment the following line:**

```
;register => 4023123982:4023123982@192.168.1.1/4023123982

[4023123982]
type=friend
host=dynamic
username=4023123982
canreinvite=no
nat=no
context=phones
disallow=all
allow=ulaw

[4023123983]
type=friend
host=dynamic
username=4023123983
canreinvite=no
nat=no
context=phones
disallow=all
allow=ulaw
```

```
[4023123984]
type=friend
host=dynamic
username=4023123984
canreinvite=no
nat=no
context=phones
disallow=all
allow=ulaw

[4023123985]
type=friend
host=dynamic
username=4023123985
canreinvite=no
nat=no
context=phones
disallow=all
allow=ulaw

[wlg-gateway]
type=friend
disallow=all
allow=ulaw
context=phones
host=192.168.1.1
canreinvite=no
nat=no
dtmfmode=rfc2833
```

**A note about SIP registrations from the Asterisk PBX:**

The User ID configured from the Optimum Business SIP Trunk Adaptor's SIP Trunk Configuration page may be any alphanumeric string (i.e.: admin, 4023123982 or abc123). Once configured, the Optimum Business SIP Trunk Adaptor will look for a matching User ID from the "Contact" header of the SIP registration request from the Asterisk PBX. If no match is found, the Optimum Business SIP Trunk Adaptor will return a 403 message. If the SIP registration request does not have a "Contact" header, then the Optimum Business SIP Trunk Adaptor will attempt to match the User ID from the "From" header.

Assuming the User ID (and password) are both configured as "4023123982", the following statement in sip.conf allows the Asterisk PBX to send the SIP Registration request with the same User ID in both the "Contact" header and the "From" header and register successfully:

```
register => 4023123982:4023123982@192.168.1.1/4023123982
```

The following is a SIP trace taken from the Optimum Business SIP Trunk Adaptor's LAN interface, an example of a successful registration from Asterisk when the User ID is configured as "4023123982":

```
# tcpdump -ni any port 5060
tcpdump: listening on any
21:42:17.612532 192.168.1.166.5060 > 192.168.1.1.5060:
>>>>>>>>>>>>>>>sip header start>>>>>>>>>>>>>>>>>>>
REGISTER sip:192.168.1.1 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.166:5060;branch=z9hG4bK5525e6a2;rport
From: <sip:4023123982@192.168.1.1>;tag=as150a65b5
To: <sip:4023123982@192.168.1.1>
Call-ID: 5dbf6b3a53635c0732df8f554a6cf235@192.168.1.166
CSeq: 102 REGISTER
User-Agent: Asterisk PBX
Max-Forwards: 70
Expires: 120
Contact: <sip:4023123982@192.168.1.166>
Event: registration
Content-Length: 0

<<<<<<<<<<<<<<<sip header stop<<<<<<<<<<<<<<<<<<<<
 (DF)
21:42:17.614949 192.168.1.1.5060 > 192.168.1.166.5060:
>>>>>>>>>>>>>>>sip header start>>>>>>>>>>>>>>>>>>>
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.1.166:5060;branch=z9hG4bK5525e6a2;rport
From: <sip:4023123982@192.168.1.1>;tag=as150a65b5
To: <sip:4023123982@192.168.1.1>
Call-ID: 5dbf6b3a53635c0732df8f554a6cf235@192.168.1.166
CSeq: 102 REGISTER
Contact: <sip:4023123982@192.168.1.166>;expires=120
Content-Length: 0
```

## Asterisk configuration file for PBX registration mode: extensions.conf

Once the PBX has successfully registered with the Optimum Business SIP Trunk Adaptor, all outbound calls sent to the Optimum Business SIP Trunk Adaptor from the PBX must also have the same User ID in the "Contact" header (or in the "From" header if the "Contact" header is missing) of the Invite. Since all the phones may have different calling numbers, the following example for dial-out rules in extensions.conf should allow Asterisk to send all calls/Invite to the Optimum Business SIP Trunk Adaptor with the "Contact" header and the "From" header set to the same as the configured User ID, 4023123982:

In Registration mode only, the Pilot DID can be sent for outbound Caller ID. Configure the User ID with the Pilot DID for each line. Note that the Caller ID for any outbound calls must be the same as the Username configured in the Optimum Business SIP Trunk Adaptor.

```
exten => _91NXXNXXXXXX,1,Set(CALLERID(num)=4023123982)
exten => _91NXXNXXXXXX,n,Dial(SIP/${EXTEN:1}@wlg-gateway,20,rt)
exten => _91NXXNXXXXXX,n,Hangup
```

**Note:**
**Many Asterisk distributions recommend moving customization parameters to extra files that are not included when the base distribution files get loaded. The typical convention is to use the same filename as the base file concatenated to "_custom.conf." For example, for modifications of extensions.conf, the Asterisk Administrator would make edits to a file named extensions_custom.conf. Different distributions of Asterisk may or may not follow this convention.**

```
; extensions.conf
;----------------
[general]
static=yes
writeprotect=no

[default]
; default context

[phones]
exten => 411,1,Dial(SIP/411@wlg-gateway,20)
exten => 4023123982,1,Dial(SIP/4023123982,15,rt)
exten => 4023123983,1,Dial(SIP/4023123983,15,rt)
exten => 4023123984,1,Dial(SIP/4023123984,15,rt)
exten => 4023123985,1,Dial(SIP/4023123985,15,rt)
exten => 3123982,1,Dial(SIP/4023123982,15,rt)
exten => 3123983,1,Dial(SIP/4023123983,15,rt)
exten => 3123984,1,Dial(SIP/4023123984,15,rt)
exten => 3123985,1,Dial(SIP/4023123985,15,rt)
exten => 3982,1,Dial(SIP/4023123982,15,rt)
exten => 3983,1,Dial(SIP/4023123983,15,rt)
exten => 3984,1,Dial(SIP/4023123984,15,rt)
exten => 3985,1,Dial(SIP/4023123985,15,rt)
```

**;Dial-out rules for Asterisk PBX with SIP registration:**

```
;dial 9 to call 10-digit number outside PBX
exten => _91NXXNXXXXXX,1,Set(CALLERID(num)=4023123982)
exten => _91NXXNXXXXXX,n,Dial(SIP/${EXTEN:1}@wlg-gateway,20,rt)
exten => _91NXXNXXXXXX,n,Hangup
exten => _9NXXNXXXXXX,1,Set(CALLERID(num)=4023123982)
exten => _9NXXNXXXXXX,n,Dial(SIP/${EXTEN:1}@wlg-gateway,20,rt)
exten => _9NXXNXXXXXX,n,Hangup

;dial 9 to call 7-digit number (within the same exchange) outside PBX
exten => _9312XXXX,1,Set(CALLERID(num)=4023123982)
exten => _9312XXXX,n,Dial(SIP/${EXTEN:1}@wlg-gateway,20,rt)
exten => _9312XXXX,n,Hangup

[from-wlg-gateway]
exten => 4023123982,1,Dial(SIP/4023123982,15,rt)
exten => 4023123983,n,Dial(SIP/4023123983,15,rt)
exten => 4023123984,n,Dial(SIP/4023123984,15,rt)
exten => 4023123985,n,Dial(SIP/4023123985,15,rt)
exten => i,n,Congestion()
```

## Asterisk configuration file for static or non-PBX registration: extensions.conf

In non-registration (static) mode, you may have any DID on the SIP Trunk configured as the Caller ID of the outbound call.

```
; extensions.conf
;----------------
[general]
static=yes
writeprotect=no

[default]
; default context

[phones]
exten => 411,1,Dial(SIP/411@wlg-gateway,20)
exten => 4023123982,1,Dial(SIP/4023123982,15,rt)
exten => 4023123983,1,Dial(SIP/4023123983,15,rt)
exten => 4023123984,1,Dial(SIP/4023123984,15,rt)
exten => 4023123985,1,Dial(SIP/4023123985,15,rt)
exten => 3123982,1,Dial(SIP/4023123982,15,rt)
exten => 3123983,1,Dial(SIP/4023123983,15,rt)
exten => 3123984,1,Dial(SIP/4023123984,15,rt)
exten => 3123985,1,Dial(SIP/4023123985,15,rt)
exten => 3982,1,Dial(SIP/4023123982,15,rt)
exten => 3983,1,Dial(SIP/4023123983,15,rt)
exten => 3984,1,Dial(SIP/4023123984,15,rt)
exten => 3985,1,Dial(SIP/4023123985,15,rt)
```

**;Dial-out rules for Asterisk PBX with no SIP registration**

```
;dial 9 to call 10-digit number outside PBX
exten => _91NXXNXXXXXX,1,Dial(SIP/${EXTEN:1}@wlg-gateway,20,rt)
exten => _91NXXNXXXXXX,n,Hangup
exten => _9NXXNXXXXXX,1,Dial(SIP/${EXTEN:1}@wlg-gateway,20,rt)
exten => _9NXXNXXXXXX,n,Hangup

;dial 9 to call 7-digit number (within the same exchange) outside PBX
exten => _9312XXXX,1,Dial(SIP/${EXTEN:1}@wlg-gateway,20,rt)
exten => _9312XXXX,n,Hangup

[from-wlg-gateway]
exten => 4023123982,1,Dial(SIP/4023123982,15,rt)
exten => 4023123983,n,Dial(SIP/4023123983,15,rt)
exten => 4023123984,n,Dial(SIP/4023123984,15,rt)
exten => 4023123985,n,Dial(SIP/4023123985,15,rt)
exten => i,n,Congestion()
```

**Note: The Asterisk PBX should populate the From field of the P-Asserted-Identity method to out pulse the DID as the calling number ID.**