SIP Trunking Configuration Guide
for
Asterisk IP-PBX
10.4.2
Table of Contents

Overview ................................................................................................................................. 2
Prerequisites .......................................................................................................................... 2
Network Topology .................................................................................................................. 4
Description of Basic Operation and Call Flows ................................................................. 5
Asterisk PBX Configuration ............................................................................................... 5
Asterisk Installation ............................................................................................................. 6
Asterisk Configuration Files ............................................................................................... 7
SIP Trunk Configuration to the EdgeMarc ........................................................................... 7
SIP Phone/Extension Configuration .................................................................................... 10
Dial plans, Auto-Attendants, and Parking lots ............................................................... 14
Parking Lot Configuration/Features Conf ........................................................................... 20
Asterisk Console Logging / Troubleshooting ................................................................. 21
SIP Trunking using the EdgeMarc Network Services Gateway and the Asterisk PBX

Overview

The purpose of this configuration guide is to describe the steps needed to configure the Asterisk PBX for proper operation in a SIP Trunking application with the e-SBC EdgeMarc. Please note that this guide documents the basic configuration needed in the Asterisk PBX and that the requirements of specific SIP Trunking environments may require modifications to the configuration steps provided in this document.

Prerequisites

SIP Trunking information provided by the VoIP service provider:

- SIP proxy server IP address or DNS name.
- Trunking Direct Inward Dial (DID) phone numbers
  - Calls to the Trunking DID(s) are forwarded from the service provider to the wide area network (WAN) IP address of the EdgeMarc. There may be a single “Pilot” phone number used for all inbound calls and/or multiple DIDs depending on the service provider settings.
- SIP authentication credentials (optional)
  - Some SIP Trunking service providers require a unique username and password to be supplied for IP PBX registrations and/or SIP signaling using P-Asserted-Identity (RFC 3325). This configuration guide provides the configuration steps for both PBX registration and static or non-registration modes of PBX operation.
Network Topology

The PBX in this network topology represents the Asterisk PBX that is connected to the LAN port of the EdgeMarc Network Services gateway. The PBX used in the lab comprises of the following:

- Asterisk PBX software version 10.4.2
- 3 Polycom SIP phones

Table 1 – PBX Information

<table>
<thead>
<tr>
<th></th>
<th>Open Source Asterisk</th>
</tr>
</thead>
<tbody>
<tr>
<td>Manufacturer:</td>
<td></td>
</tr>
<tr>
<td>Model:</td>
<td>SIP</td>
</tr>
<tr>
<td>Software Version:</td>
<td>10.4.2</td>
</tr>
<tr>
<td>Does the PBX send SIP</td>
<td>Yes</td>
</tr>
<tr>
<td>Registration messages</td>
<td></td>
</tr>
<tr>
<td>(Yes/No)?</td>
<td></td>
</tr>
<tr>
<td>Vendor Contact:</td>
<td>Irc.freenode.net #asterisk</td>
</tr>
</tbody>
</table>
**Table 2 – E-SBC Information**

<table>
<thead>
<tr>
<th>Manufacturer</th>
<th>Edgewater Network, Inc.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Model</td>
<td>EdgeMarc 4550</td>
</tr>
<tr>
<td>Software Version</td>
<td>11.6.13</td>
</tr>
</tbody>
</table>

**Description of Basic Operation and Call Flows**

**Basic Call Flow:**

All LAN phones are connected to the Asterisk PBX. The Asterisk will interface with the service provider using a SIP trunk to the EdgeMarc.

Internal calls:
- Calls between phones on the LAN side of the EdgeMarc/PBX
- LAN phone → Asterisk → LAN phone

Outbound calls:
- Call is initiated by a LAN phone to a WAN phone.
- LAN phone → Asterisk <SIP trunk> → EdgeMarc → SIP trunk service provider → WAN phone

Inbound call:
- Call is initiated by a WAN phone to a LAN phone.
- WAN phone → SIP trunk service provider → EdgeMarc → <SIP trunk> Asterisk → LAN phone

**Asterisk PBX Configuration**

The steps below describe the basic configuration required to enable the Asterisk PBX to use a SIP trunk for inbound and outbound calling. Please refer to the Asterisk documentation for other advanced PBX features.

This configuration is based on Asterisk software version 10.4.2, however versions 1.8.7 and 1.8.12 were setup and tested with basic call flows.
Many other PBX systems are based off of the Asterisk core, such as Trixbox, FreePBX, and AsteriskNOW as well as many others. Even though they may have GUI's, the core configuration files and syntax is the same, and the core functions and operations are the same. Just take into account that like most other UNIX/Linux programs, the core configuration file directories can be placed anywhere in the system.

This guide will go through the steps of installing Asterisk 10.4.2/1.8.7/1.8.12 on a Linux/GNU OS. Explain the configuration files and their specific purposes. Configuration of 3 phones on the LAN side, configuration of the inbound and outbound trunks to the EdgeMarc. Configuration of the dial plans, Auto-Attendants, and parking lots, as well as basic console troubleshooting for the Asterisk system.

**Asterisk Installation**

The method of installation for Asterisk is based on the OS being used to host the Asterisk PBX. The easiest method is to use the OS's built in package installer, such as yum or apt-get.

For the Fedora/ CentOS/ RHEL/ SuSE OS's:

```
# yum -y install asterisk asterisk-sounds
```

For Debian/Ubuntu/ Mint OS's:

```
apt-get install asterisk asterisk-sounds
```

These steps will install the Asterisk software from their latest supported repository.

If a binary install is not possible, compiling from the Asterisk source code is another option.

Download and install the source code:

```
# cd /usr/src/asterisk
# wget http://downloads.asterisk.org/pub/telephony/asterisk/releases/asterisk-10.4.2.tar.gz
# tar xfvz asterisk-10.4.2.tar.gz
# cd asterisk-10.4.2/
# ./compile
# make clean
# make
# make install
```
Asterisk Configuration Files

As part of this configuration guide there will be 3 conf files that will be explained and configured. The extensions.conf file, the sip.conf file, and the features.conf file.

Extensions.conf

This conf file contains the extensions for phones and DID's the come into the PBX, dial plans, Auto-attendants, and call forwarding configurations.

Sip.conf

This conf file contains the global register configuration to the SIP trunks, the inbound and outbound call settings, and the phone/extension configuration and registration settings.

Features.conf

This is where the parking lot is configured.

SIP Trunk Configuration to the EdgeMarc

Within the sip.conf file resides the configuration for working with the SIP Trunk. All configurations in this file must go under the [General] section.

Add the register string, this is only required if the Asterisk PBX needs to register to the EdgeMarc or SIP Provider directly. The register string MUST come before any phone/extension or trunk configuration, and directly after the [General] section.

Format:
register => user:secret:@host[:port]/extension

Example register string:
register => 6785551234:p@ssw0rd@192.168.1.1:5060/6785551234

The DID listed here, 6785551234 is the pilot DID of the SIP Trunk Group, it is the Authentication Username that the EdgeMarc looks for when a registration originates from the PBX. This can be changed to anything as long as the EdgeMarc is changed to reflect these setting. Following it is a ":" to signify the next part of the registration parameters.
The next part is the Authentication Password the EdgeMarc looks for when the PBX registers to the EdgeMarc. Next is the IP/domain of the SIP server, the LAN side IP address of the EdgeMarc acts as the SIP server to the PBX, @192.168.1.1. The extension/DID that is attached to the end of the string is to indicate what user uri to use, by default it will be “s” if this is not defined.

This string will cause a register attempt to the EdgeMarc with the Authentication Username of 6785551234 and password of p@ssw0rd to the SIP server address of 192.168.1.1. It attempts to register every 60 seconds by default.

Name the inbound trunk section:

[Inbound]

State what type of connection this is. A SIP trunk is a “peer” in this instance:

type=peer

State the DTMF method to be used:

dtmfmode=rfc2833

There are 4 options for DTMF; inband, RFC2833, SIP INFO, and auto, which will use whatever method is negotiated from the SIP Provider.

dtmfmode=auto
dtmfmode=inband
dtmfmode=rfc2833
dtmfmode=info

The inband option requires that ONLY the G.711 codec be used for audio streams (explained further down). The RFC2833 option is the recommended and most commonly used option as it uses RTPEVENTs as the signaling method. The SIP info is not as common and requires that the SIP Provider be capable of accepting and converting this DTMF method into inband or RFC2833.

Define which codecs to be used with the audio streams, first disallow ALL codecs:

disallow=all

Then follow with the allowed codecs:

allow=ulaw
allow=g729
allow=alaw
Ulaw and alaw is G.711.

Specify how to handle connections with peers:

```
insecure=invite, port
```

Select a context name for your Inbound Trunk to reference to, this is case sensitive and is referred to in other conf files:

```
context=inbound
```

A very important option is to tell Asterisk if it is behind a NAT or if it is not behind a NAT. Even though the EdgeMarc is NAT'ing the IP headers to and from Asterisk, the VoIP ALG built into the EdgeMarc will deal with the proper header manipulations for SIP. Turn off NAT in the Asterisk to prevent header manipulation conflicts:

```
nat=no
```

Specify the IP from which the SIP signaling will come from, this is the same as the SIP Server, set it to the EdgeMarc's LAN IP address:

```
host=192.168.1.1
```

Set the port for Asterisk to bind/listen to:

```
bindport=5060
```

That's it for the [Inbound] Trunk section.

For the Outbound Trunk Section:

Set the section name:

```
[Outbound]
```

Specify if the Outbound Trunk/peer should send SIP OPTION messages (keep alives) to the SIP Server/Edgemarc to check and see if the SIP server is there and active, this is only useful if multiple outbound trunks are used, turn this off:

```
qualify=no
```

Set the connection type:

```
type=peer
```
Set the DTMF method to be used on outbound tones:

dtmfmode=rfc2833

Set the audio codecs to be used and negotiated on outbound calls:

disallow=all
allow=ulaw
allow=alaw
allow=g729

Specify the domain that will be used in the FROM and TO headers on outbound calls:

fromdomain=docsis.cox.net

Set the context, remember this is case sensitive and will be referenced later on:

context=outbound

Specify the Caller ID, the Caller ID Number, and the FROM user which will show up in the FROM field of all SIP messages originating from the Asterisk. The callerid can be set to anything within 16 characters, this is usually the name that shows up:

callerid=Anonymous

or

callerid=6785551234

Set the fromuser for the FROM header:

fromuser=6785551234

And the Caller ID Number:

cid_number=6785551234

Note: When an extension has a DID specified as the callerID/fromuser/cid_number, these last three options will need to be removed to prevent overriding the DID assigned to the extension.

**SIP Phone/Extension Configuration**

The phone/extension configuration resides in the sip.conf file, directly after the trunk/peers configuration.

Name the phone/extension. Typically this is the extension assigned to the phone but it can also be a name:
Specify the type of connection, a phone/extension or device on the LAN side of the PBX is called a “friend” in this instance:

```
type=friend
```

Specify the IP address the phone/extension is at, most of the time phones get IP addresses from a DHCP server, if that's the case, set the host option to dynamic:

```
host=dynamic
```

Set the context of the phone/extension, just like the trunks/peers this is case sensitive and referenced to later on:

```
context=phones
```

Set the registration username the phone/extension will use to register to the PBX:

```
username=100
```

Set the password the phone/extension will use to register to the PBX (note, it is CRITICAL to set a strong password to warn off hacking attempts):

```
secret=p@ssw0rd
```

Set the dtmf method used by the phone:

```
dtmfmode=rfc2833
```

Set the NAT options for the phone/extension:

```
nat=no
```

Set the audio stream codecs for the phone/extension to negotiate:

```
disallow=all
allow=ulaw
allow=alaw
allow=g729
```

Set the allowed IP range where the phone/extension is permitted to come from, this is a good security step to prevent fraud:

```
deny=0.0.0.0/0
permit=192.168.1.0/255.255.255.0
```
If this particular phone/extension will be attached to a DID, set the caller ID, caller ID number, and the FROM name to be used (if these options are set in the Outbound trunk section, the Outbound trunk settings will override the phone caller ID settings):

```
cid_number=6785551233
callerid=6785551233
fromuser=6785551233
```

Below is how it should look in the sip.conf file:

```
[100]
type=friend
host=dynamic
context=phones
username=100
secret=p@ssw0rd
dtmfmode=auto
nat=no
disable=all
allow=ulaw
allow=alaw
allow=g729
deny=0.0.0.0/0
permit=192.168.1.0/255.255.255.0
cid_number=6785551233
callerid=6785551233
fromuser=6785551233
```

Repeat for other phones/extensions.

This completes the configuration of the sip.conf file. This is how it should look:

```
[General]
register => 6785551234:p@ssw0rd@192.168.1.1:5060

[Inbound]
type=peer
dtmfmode=auto
disable=all
allow=ulaw
allow=g729
allow=alaw
insecure=invite,port
```
context=inbound
nat=no
host=192.168.1.1
bindport=5060

[Outbound]
qualify=no
type=peer
dtmfmode=rfc2833
disallow=all
allow=ulaw
allow=g729
allow=alaw
fromdomain=docsis.cox.net
insecure=invite,port
context=outbound
nat=no
host=192.168.1.1
cid_number=6785551234
callerid=Anonymous
fromuser=6785551234

[100]
type=friend
host=dynamic
context=phones
username=100
secret=p@ssw0rd
dtmfmode=auto
nat=no
disallow=all
allow=ulaw
allow=alaw
allow=g729
deny=0.0.0.0/0
permit=192.168.1.0/255.255.255.0
clid_number=6785551233
callerid=6785551233
fromuser=6785551233

[Bob]
type=friend
host=dynamic
context=phones
username=101
secret=p@ssw0rd
dtmfmode=auto
nat=no
disallow=all
allow=ulaw
allow=alaw
allow=g729
deny=0.0.0.0/0
permit=192.168.1.0/255.255.255.0
cid_number=6785551232
callerid=6785551232
fromuser=6785551232

[Fax]
type=friend
host=dynamic
context=phones
username=102
secret=p@ssw0rd
dtmfmode=auto
nat=no
disallow=all
allow=ulaw
allow=alaw
allow=g729
deny=0.0.0.0/0
permit=192.168.1.0/255.255.255.0
cid_number=6785551231
callerid=6785551231
fromuser=6785551231

**Dial plans, Auto-Attendants, and Parking lots**

Like the sip.conf file, extensions.conf needs to start with a [General] section, everything else will be configured after this section.

[General]
static=yes
writeprotect=no
clearglobalvars=no

To define the DID's that will come into the Asterisk from the SIP Server, use the context defined in the sip.conf file under the Inbound Trunk/peer section:

[inbound]

Tell asterisk what to do with the DID's when they come from the SIP Trunk:
The first line, when a call comes into the Asterisk from the Inbound Trunk, and it is destined to 6785551234, answer it automatically. Second line, for the same number, send it to aa,s,1. “aa” is the Auto-Attendant that is setup further down. Third line simply says Hangup the call once the Auto-Attendant is done processing the call.

Notice the 1, then below that the n and then below that another n. This is the numbering scheme for the call flow. A 2, then a 3, could also be used.

Set up another DID, this time sending it to a phone/extension configured in the sip.conf file.

Same flow as before, except this time Dial SIP extension 100.

Repeat for all DID’s that the Asterisk will use.

Configure each phones extension, this is required so that Asterisk knows how each phone/extension can talk to each other, or where the phones/extensions are at to forward calls to them, make sure this section is named the same as the context section that the phones/extensions are under:

[phones]

Property of Cox Communications, Inc.  Version 0.2
An extension may have been named Bob or Fax, but the 3 digit extension is dialed, not the name of the extension. However the name is what's referenced when Asterisk looks for it.

If a phone/extension makes an outbound call, Asterisk needs to know which trunk to send the call out through, it also needs to know which dialed digits are allowed.

Look at a single digit, if “0” was to be dialed to get an operator and sent to the Outbound Trunk/peer:

```
  exten => _X,1,Answer()
  exten => _X,n,Log(NOTICE, Dialing out from ${CALLERID(all)} to ${EXTEN} through Outbound)
  exten => _X,n,Dial(SIP/Outbound/${EXTEN})
  exten => _X,n,Playtones(congestion)
  exten => _X,n,Hangup()
```

The first line tells Asterisk to grab the call, it “Answers” the call. The second line says log the call to the console, this will be shown in the Asterisk console for debugging. The Third line says send the call to the Outbound trunk/peer section specified in the sip.conf file, this is NOT the outbound context, but rather the section name. The fourth line is play a “Not Available” tone if the call was not sent to the Outbound Trunk/peer, or if the trunk/peer rejected the call. The final line Hangs up the call.

Repeat for all dial plans, Single digit( 0 for the operator), 2 digits( 00 for the operator), 3 digit(911, 311, 411), local calls(xxx-xxxx, 555-1111), long distance(xxx-xxx-xxxx, 677-555-1111), long distance with 1(1-677-555-1111), and international (011).

Also add a line under this section for the parking lot, this will be explained later:

```
include => parkedcalls
```

Setup the Auto-Attendant.

```
[aa]
  exten => s,1,Answer
  exten => s,n,Background(/var/lib/asterisk/sounds/PleaseDialExt)
  exten => s,n,WaitExten
  exten => 100,1,Dial(SIP/100)
  exten => 100,n,Hangup
  exten => 101,1,Dial(SIP/Bob)
  exten => 101,n,Hangup()
```

Line 1 picks up calls sent to the aa context.

Line 2 plays the sound file PleaseDialExt, which is located in /var/lib/asterisk/sounds, while it is being played, the Auto-Attendant is waiting for input from the caller.
Line 3 tells the auto-attendant to wait until there is a response from the caller. Line 4 is if the user hits 100, forward the caller to extension 100. Line 6 is the same as line 4, but for extension 101 (Bob).

Set up call forwarding.

[apps]

; Unconditional Call Forward
exten => _*21*X.,1, DBput (CFIM/\${CALLERIDNUM}=\${EXTEN:4})
exten => _*21*X.,2, Hangup
exten => #21#,1, DBdel (CFIM/\${CALLERIDNUM})
exten => #21#,2, Hangup

; Call Forward on Busy or Unavailable
exten => _*61*X.,1, DBput (CFBS/\${CALLERIDNUM}=\${EXTEN:4})
exten => _*61*X.,2, Hangup
exten => #61#,1, DBdel (CFBS/\${CALLERIDNUM})
exten => #61#,2, Hangup

The first 4 lines define Call Forward Always, this is setup when a phone/extension dials *21* then the number to forward the caller to. #21# removes the forward always.

The next 4 lines defines when callers should be forwarded only when the phone/extension being called is busy, this is setup when a phone/extension dials *61* then the number to forward the caller to. #61# removes the forward busy option.

This completes the extensions.conf config file. Below is the completed config file:

[General]
static=yes
writeprotect=no
clearglobalvars=no

[inbound]
exten => _6785551234,1, Answer
exten => _6785551234,n, GoTo(aa,s,1)
exten => _6785551234,n, Hangup()

exten => _6785551233,1, Answer
exten => _6785551233,n, Dial(SIP/100)
exten => _6785551233,n, Hangup()
exten => _6785551232,1,Answer
exten => _6785551232,n,Dial(SIP/Bob)
exten => _6785551232,n,Hangup()

exten => _6785551231,1,Answer
exten => _6785551231,n,Dial(SIP/Fax)
exten => _6785551231,n,Hangup()

[phones]

exten => 100,1,Dial(SIP/100)
exten => 100,n,Hangup()

exten => 101,1,Dial(SIP/Bob)
exten => 101,n,Hangup()

exten => 102,1,Dial(SIP/Fax)
exten => 102,n,Hangup()

exten => _X,1,Answer()
exten => _X,n,Log(NOTICE, Dialing out from ${CALLERID(all)} to ${EXTEN} through Outbound)
exten => _X,n,Dial(SIP/Outbound/${EXTEN})
exten => _X,n,Playtones(congestion)
exten => _X,n,Hangup()

exten => _XX,1,Answer()
exten => _XX,n,Log(NOTICE, Dialing out from ${CALLERID(all)} to ${EXTEN} through Outbound)

exten => _XX,n,Dial(SIP/Outbound/${EXTEN})
exten => _XX,n,Playtones(congestion)
exten => _XX,n,Hangup()

exten => _XXX,1,Answer()
exten => _XXX,n,Log(NOTICE, Dialing out from ${CALLERID(all)} to ${EXTEN} through Outbound)
exten => _XXX,n,Dial(SIP/Outbound/${EXTEN})
exten => _XXX,n,Playtones(congestion)
exten => _XXX,n,Hangup()
exten => _XXXX,1,Answer()
exten => _XXXX,n,Log(NOTICE, Dialing out from ${CALLERID(all)} to
${EXTEN}) through Outbound
exten => _XXXX,n,Dial(SIP/Outbound/${EXTEN})
exten => _XXXX,n,Playtones(congestion)
exten => _XXXX,n,Hangup()

exten => _NXXXXXX,1,Answer()
exten => _NXXXXXX,n,Log(NOTICE, Dialing out from ${CALLERID(all)} to
${EXTEN}) through Outbound
exten => _NXXXXXX,n,Dial(SIP/Outbound/${EXTEN})
exten => _NXXXXXX,n,Playtones(congestion)
exten => _NXXXXXX,n,Hangup()

exten => _NXXNXXXXXX,1,Answer()
exten => _NXXNXXXXXX,n,Log(NOTICE, Dialing out from ${CALLERID(all)} to
${EXTEN}) through Outbound
exten => _NXXNXXXXXX,n,Dial(SIP/Outbound/${EXTEN})
exten => _NXXNXXXXXX,n,Playtones(congestion)
exten => _NXXNXXXXXX,n,Hangup()

exten => _1NXXNXXXXXX,1,Answer()
exten => _1NXXNXXXXXX,n,Log(NOTICE, Dialing out from ${CALLERID(all)} to
${EXTEN}) through Outbound
exten => _1NXXNXXXXXX,n,Dial(SIP/Outbound/${EXTEN})
exten => _1NXXNXXXXXX,n,Playtones(congestion)
exten => _1NXXNXXXXXX,n,Hangup()

exten => _011.,1,Answer()
exten => _011.,n,Log(NOTICE, Dialing out from ${CALLERID(all)} to
${EXTEN}) through Outbound
exten => _011.,n,Dial(SIP/Outbound/${EXTEN})
exten => _011.,n,Playtones(congestion)
exten => _011.,n,Hangup()

include => parkedcalls

[aa]

exten => s,1,Answer
exten => s,n,Background(/var/lib/asterisk/sounds/PleaseDialExt)
exten => s,n,WaitExten
exten => 100,1,Dial(SIP/100)
exten => 100,n,Hangup
exten => 101,1,Dial(SIP/Bob)
exten => 101,n,Hangup()

[apps]

; Unconditional Call Forward
exten => _*21*X.,1,DBput(CFIM/${CALLERIDNUM}=${EXTEN:4})
exten => _*21*X.,2,Hangup
exten => #21#,1,DBdel(CFIM/${CALLERIDNUM})
exten => #21#,2,Hangup

; Call Forward on Busy or Unavailable
exten => _*61*X.,1,DBput(CFBS/${CALLERIDNUM}=${EXTEN:4})
exten => _*61*X.,2,Hangup
exten => #61#,1,DBdel(CFBS/${CALLERIDNUM})
exten => #61#,2,Hangup

Parking Lot Configuration / features.conf

The parking lot is configured under the features.conf section.

Like the extensions.conf and sip.conf files, features.conf needs to start with a [General] section and everything needs to be configured after this point.

[General]

Now put in the configuration for the parking lot:

parkext => 700 ; What extension to dial to park. Set per parking lot.
parkpos => 701-720 ; What extensions to park calls on.
(default parking lot)
context => parkedcalls ; Which context parked calls are in
(default parking lot)
parkingtime => 15 ; Number of seconds a call can be parked before returning.
comebacktoorigin = yes ; Setting this option configures the behavior of call parking when the
courtesytone = beep ; Sound file to play to when someone picks up a parked call
**Asterisk Console Logging / Troubleshooting**

To enable/enter the asterisk console:

```
# asterisk -rvvvvvv
```

The number of v's indicates the verbosity of logging, 7 is the max.

The asterisk console can show the sip signaling that occurs between the phones and the SIP trunks, to enable this sip logging:

```
*CLI> sip set debug on
```

If changes were made to the conf files, reload the plans:

```
*CLI> reload
```

For advanced configurations and support please contact the Edgewater Technical Assistance Center support@edgewaternetworks.com or call 408.351.7255.