

# Getting Started With Open Source Telephony

## A Beginners Guide to Asterisk

**Steve Sokol**

Asterisk Marketing Lead, Digium

**Justin Hester**

Lead Asterisk Technical Instructor, Digium

- **Summary of Asterisk and Distributions**
  - Asterisk is a toolkit
  - Distros are more complete package with GUI
- **Getting started using Asterisk**
  - Architecture - Linux + Asterisk
  - Difference between CLI and GUI
  - Versioning
- Basics
- Resources
- Training classes
- **Digium and Asterisk**
- **AstriCon 2016**



# Getting Started with Asterisk

# Getting Started with Asterisk

- Find it
- Install it
- Configure it

## An **Open Source** Communications Platform

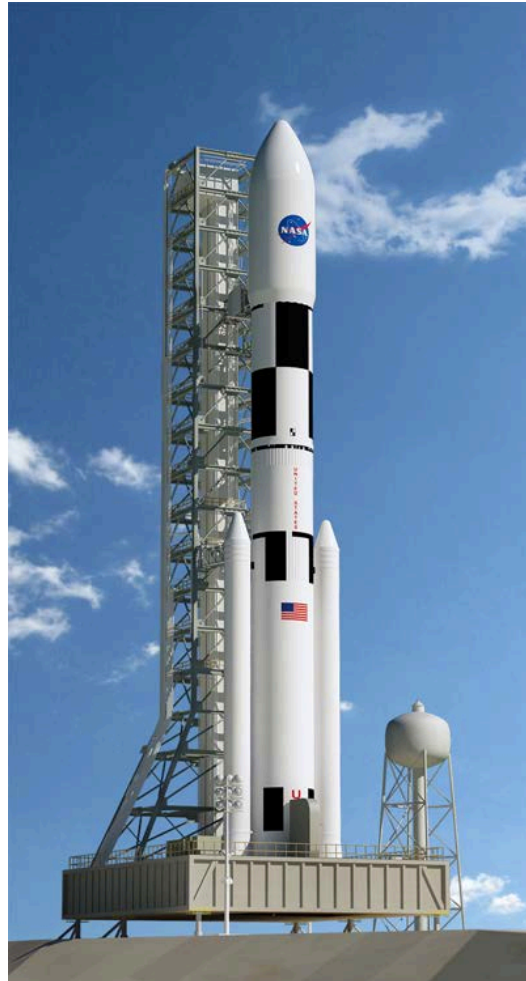
- Software, written in C, that you put on an ordinary operating system—transforming that system into a communications engine.

# Software – A communications platform

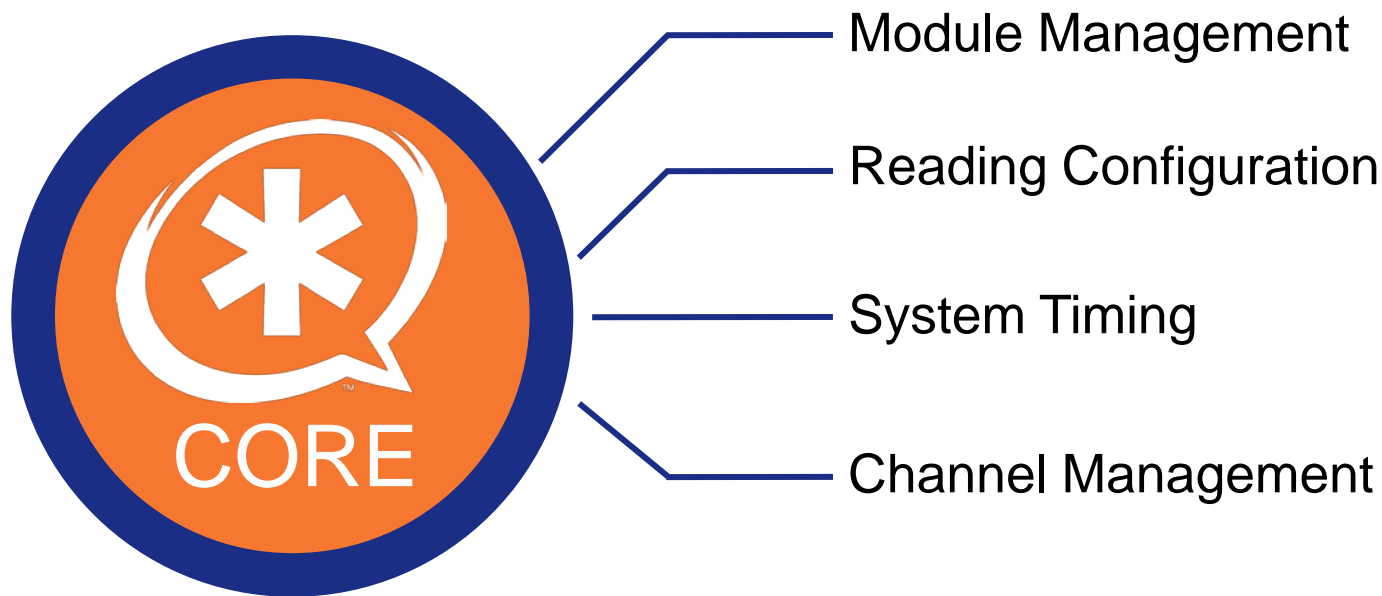
A system through which **communications** flow, from one endpoint to another.



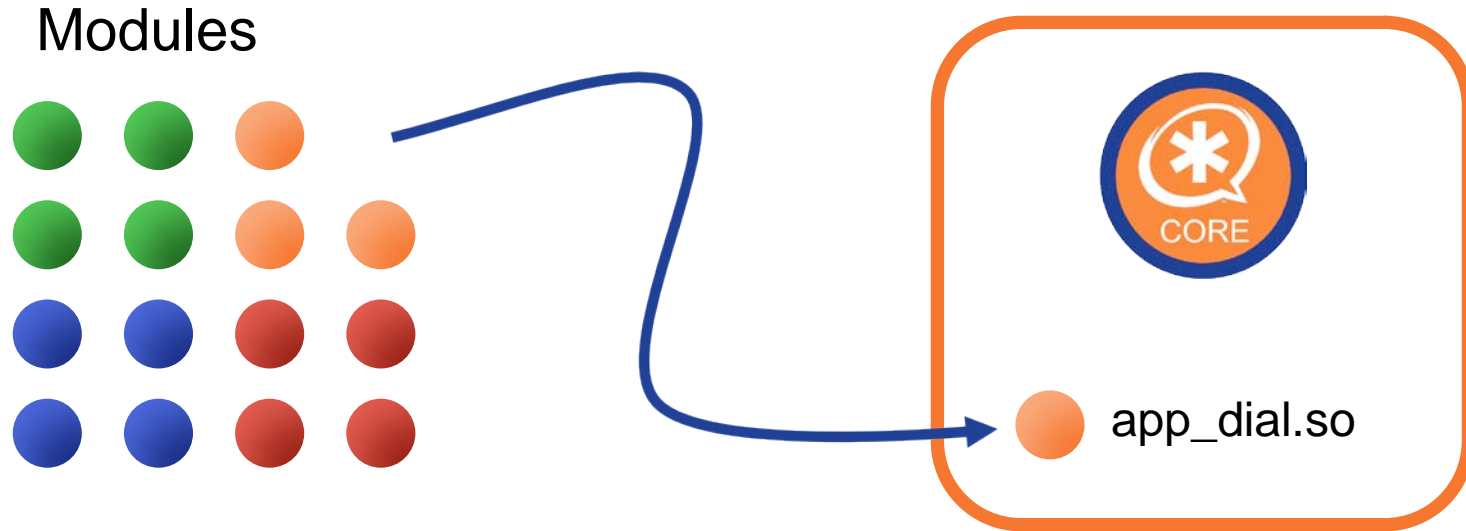
## Open Source Communications **Platform**



A simple core with only a few responsibilities







- Use the native modules
- Use Asterisk's APIs to control and extend Asterisk – AMI, AGI, and ARI

# Preparing for Asterisk – Set up a host machine

- Old physical hardware
  - Laptop, rackmount or tower system
- Virtual machine
  - e.g. VirtualBox (on your Mac, Windows or Linux laptop!) [www.virtualbox.org](http://www.virtualbox.org)
- Cloud server
  - e.g. [www.digitalocean.com](http://www.digitalocean.com) or [aws.amazon.com](http://aws.amazon.com)

# Finding Asterisk – Choose your path

## Asterisk Downloads

Download the currently supported versions of Asterisk and various Asterisk-related open source projects.



[Asterisk Communications Framework](#) • [AsteriskNOW Software PBX](#) • [DAHDI](#) • [libpri](#)

‘Source’ or  
‘Plain Vanilla’

### Asterisk Communications Framework

Asterisk is an open source toolkit for building communications applications. For a more detailed explanation, check out the [Get Started](#) section. For more information on how to use Asterisk, see the [Configuration](#) and [Operation](#) sections of the wiki.

 [Download Asterisk](#)

Latest Version - 13.3.2

[ChangeLog](#) [All Asterisk Versions](#)

AsteriskNOW,  
a PBX ‘distro’.

### AsteriskNOW Software PBX

AsteriskNOW is the premier, ready-to-run distribution of open source Asterisk. AsteriskNOW is an ISO image that installs Linux, Asterisk and the FreePBX GUI in a single, simple install. For more information, including installation instructions, check out the [AsteriskNOW](#) page.

 [AsteriskNOW 6.12 32-bit](#)

 [AsteriskNOW 6.12 64-bit](#)

Version 6.12.65-26

[www.asterisk.org/downloads](http://www.asterisk.org/downloads)

- Install a Linux operating system
- Set up networking
- Configure software repositories
- Install Asterisk dependencies
- Download and install Asterisk, DAHDI, LibPRI from provided scripts.



And you will have an unconfigured, pristine, ready to configure “Asterisk Configuration Framework”.

But it is not a PBX, or much of anything yet. . .

# Source – a glance at configuration

```
Aor: <Aor.....> <MaxContact>
Contact: <Aor/ContactUri.....> <Status....> <
RTT(ms)..>
=====
=====
Aor: 6001                                     1
                                             [dpma_endpoint]
                                             type=endpoint
ParameterName      : ParameterValue
=====
authenticate_qualify : false
contact            :
default_expiration  : 3600
mailboxes          : 6001@default
max_contacts       : 1
maximum_expiration  : 7200
minimum_expiration  : 60
outbound_proxy     :
qualify_frequency   : 0
remove_existing    : false
support_path       : false
                                             ;=====TRANSPORTS=====
                                             ;
                                             [transport-udp]
                                             type=transport
                                             protocol=udp
                                             bind=0.0.0.0
                                             ;=====ENDPOINT PHONES=====
                                             ;
                                             [endpoint_defaults1(!)
                                             type=endpoint
                                             context=localphones
                                             disallow=all
                                             allow=ulaw,gsm
asterisk13*CLI> pjsip show endpoints_      [6001]
                                             type=aor
                                             max_contacts=1
                                             mailboxes=6001@default
```

- Configure phones, endpoints, services by editing text files in Linux, or you can set up databases.
- Manage Asterisk through the “Asterisk CLI”.

# Source – example of configuring a phone

extensions.conf

```
[localphones]

exten => 6001,1,Dial(PJSIP/6001,30)
same => n,VoiceMail(6001@default)
same => n,Hangup()
```

pjsip.conf

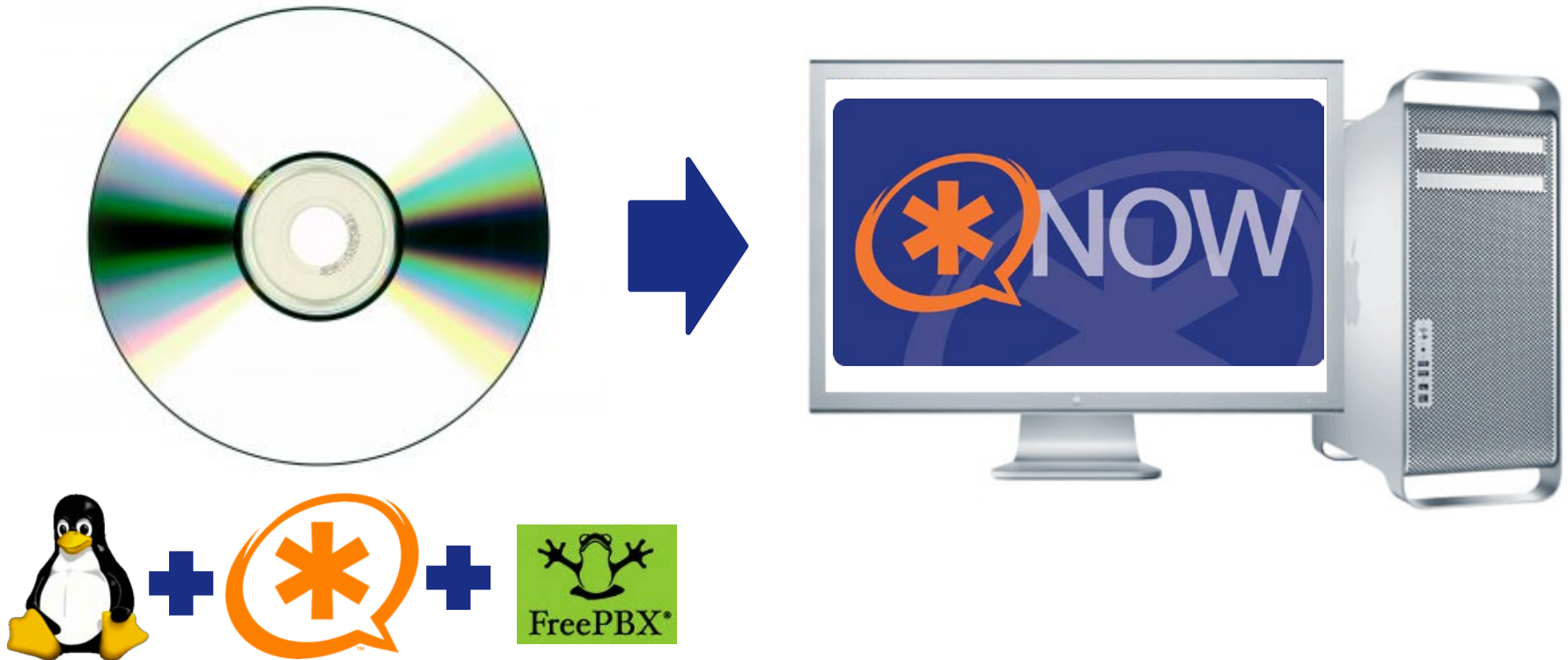
```
[6001]
type=endpoint
context=default
disallow=all
allow=ulaw
transport=simpletrans
auth=auth6001
aors=6001

[auth6001]
type=auth
auth_type=userpass
Password=MyP@ssw0rd
username=6001

[6001]
type=aor
contact=sip:6001@192.0.2.1:5060
```



# Choosing Asterisk – a distribution



- Download and install the AsteriskNOW .iso
- You'll have a PBX system for configuration.

```

Welcome to FreePBX 6.12.65
-----
FreePBX 6.12.65 with Asterisk 13
Full Install
Full Install -- No RAID
Full Install -- Advanced
HA Install -- Requires 250G or larger disk
FreePBX 6.12.65 with Asterisk 11
Full Install
Full Install -- No RAID
Full Install -- Advanced
HA Install -- Requires 250G or larger disk

```

Press [Tab] to edit options

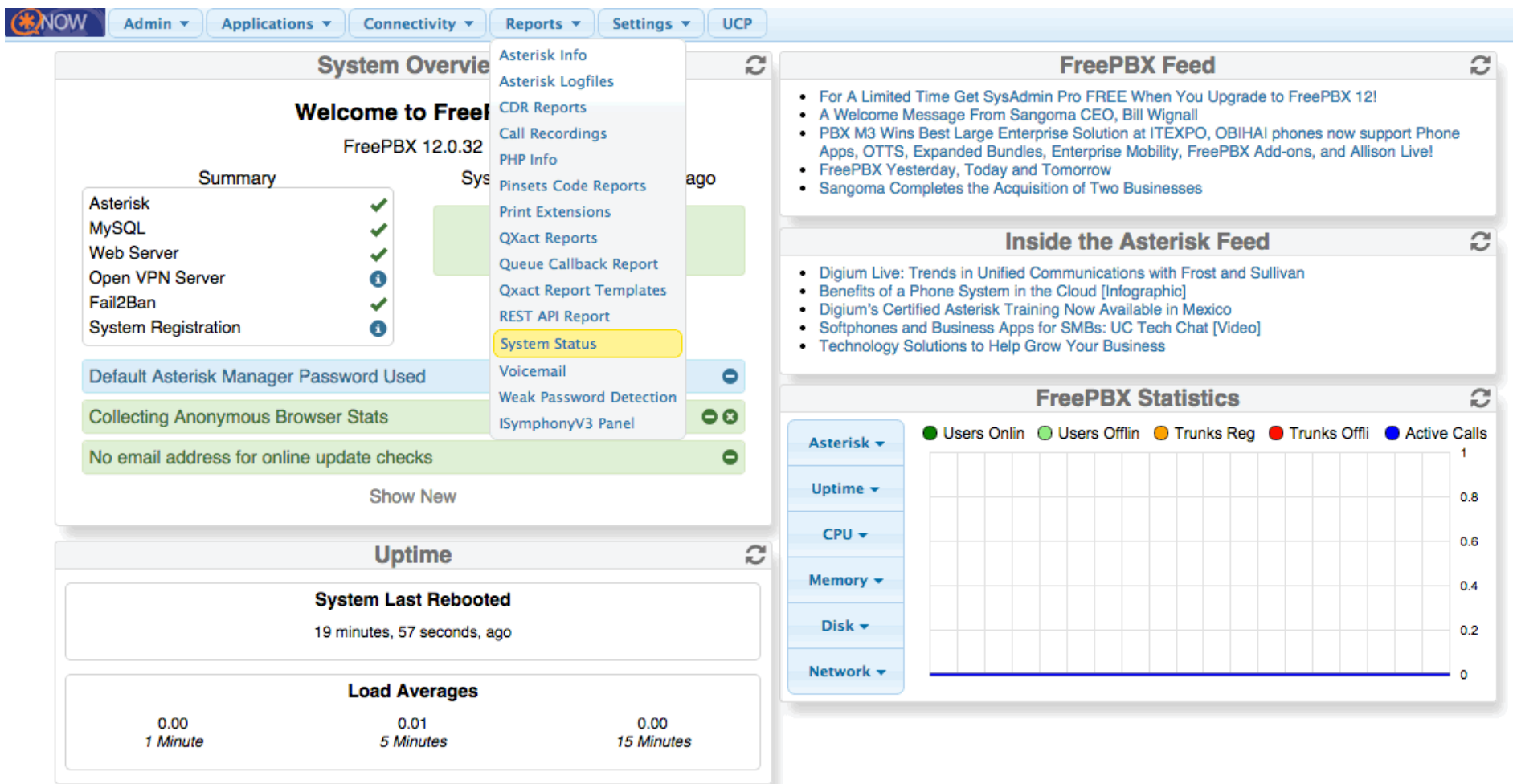


© Copyright 2014 Schmooze Com, Inc. All Rights Reserved.





# Distribution – a glance at configuration



**System Overview**

Welcome to FreePBX 12.0.32

Summary

Asterisk	✓
MySQL	✓
Web Server	✓
Open VPN Server	i
Fail2Ban	✓
System Registration	i

Default Asterisk Manager Password Used

Collecting Anonymous Browser Stats

No email address for online update checks

Show New

**Reports**

- Asterisk Info
- Asterisk Logfiles
- CDR Reports
- Call Recordings
- PHP Info
- Pinsets Code Reports
- Print Extensions
- QXact Reports
- Queue Callback Report
- Qxact Report Templates
- REST API Report
- System Status**
- Voicemail
- Weak Password Detection
- ISymphonyV3 Panel

**FreePBX Feed**

- For A Limited Time Get SysAdmin Pro FREE When You Upgrade to FreePBX 12!
- A Welcome Message From Sangoma CEO, Bill Wignall
- PBX M3 Wins Best Large Enterprise Solution at ITEXPO, OBIHAI phones now support Phone Apps, OTTS, Expanded Bundles, Enterprise Mobility, FreePBX Add-ons, and Allison Live!
- FreePBX Yesterday, Today and Tomorrow
- Sangoma Completes the Acquisition of Two Businesses

**Inside the Asterisk Feed**

- Digium Live: Trends in Unified Communications with Frost and Sullivan
- Benefits of a Phone System in the Cloud [Infographic]
- Digium's Certified Asterisk Training Now Available in Mexico
- Softphones and Business Apps for SMBs: UC Tech Chat [Video]
- Technology Solutions to Help Grow Your Business

**FreePBX Statistics**

● Users Onlin ● Users Offlin ● Trunks Reg ● Trunks Offli ● Active Calls

Asterisk ▼

Uptime ▼

CPU ▼

Memory ▼

Disk ▼

Network ▼

**Uptime**

System Last Rebooted

19 minutes, 57 seconds, ago

**Load Averages**

0.00	0.01	0.00
1 Minute	5 Minutes	15 Minutes

# Distribution – example of configuring phone



**NOW** Admin Applications Connectivity Reports Settings UCP

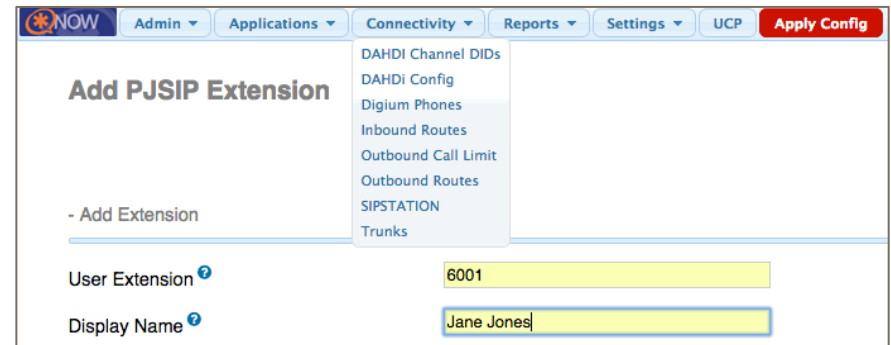
## Add an Extension

Please select your Device below then click Submit

- Device

---

Device Generic PJSIP Device



**NOW** Admin Applications Connectivity Reports Settings UCP

## Add PJSIP Extension

- DAHDI Channel DIDs
- DAHDI Config
- Digium Phones
- Inbound Routes
- Outbound Call Limit
- Outbound Routes
- SIPSTATION
- Trunks

- Add Extension

---

User Extension <sup>?</sup>

Display Name <sup>?</sup>

<http://www.asterisk.org/downloads>



Source



Distribution

## Source

<https://wiki.asterisk.org/wiki/display/AST/Beginning+Asterisk>

<http://blogs.digium.com/2015/02/24/install-asterisk-13-pjsip-centos-6/>

## Distribution

<https://wiki.asterisk.org/wiki/display/AST/Installing+AsteriskNOW>

# Asterisk – Configure it.

## Documentation & best practices

<https://wiki.asterisk.org/wiki/display/AST/Home>

<https://wiki.freepbx.org/display/FPG/Configuring+your+PBX>

## Training

<https://www.digium.com/en/training/asterisk>

## User lists and Forums (both asterisk and freepbx)

<https://community.asterisk.org/>

<http://community.freepbx.org/>

Asterisk – Go further with it.

## Community

<http://www.asterisk.org/community>

<http://www.asterisk.org/community/astricon-user-conference>

# Digium and Asterisk

- AstriCon is the annual Asterisk users conference
- Attendees will learn about
  - Trends in Asterisk use
  - The growing Asterisk ecosystem
  - The newest applications and a wide range of technical topics from Asterisk developers, users and entrepreneurs.







- **Metered Rate Plan**

- Works well for businesses that may need more flexibility in their communications week to week.
- Priced on set rate per minute of usage.
- Dynamically increase call capacity.
- Built to 10 channels by default

- **Channelized Rate Plan**

- Perfect for businesses that prefer a set, predictable monthly phone bill.
- Unlimited inbound and outbound local and long distance calls on a per channel per call basis.
- Channels can always be added for more capacity.



///////////////// Introducing ///////////////////

# Digium® Phones

Changing the way the world communicates. *Again.*



HDVoice

# Digium Phone Models



## D40/D45

Entry-level

## D50

Mid-level

## D70

Executive-level

Digium's entry-level phone with 2 line keys. This is Digium's best value phone designed for any employee in the company.

Digium's mid-level phone with 4 line keys and 10 rapid dial/busy lamp field keys for your most important contacts.

Digium's executive-level phone with 6 line keys designed for administrators and executives who need to manage up to 100 contacts.

Line Keys	2	4	6
Feature Keys	4	6	10
Rapid Dial/ Busy Lamp Field Keys	0	10 keys	10 keys - 100 contacts
Ethernet LAN and PC Port	10/100Base-T (D40) 10/100/1000Base-T (D45)	10/100Base-T	10/100/1000Base-T

# Digium VoIP Gateways



G800 Octal T1/E1/PRI  
appliance

G400 Quad T1/E1/PRI  
appliance

G200 Dual T1/E1/PRI  
appliance

G100 Single T1/E1/PRI  
appliance

- Digital TDM (T1/E1/PRI) to SIP Appliances
- Designed and built by Digium
- No moving parts
- Field proven reliable design

# Digium Telephony Cards

- **Full line of PCI-E and PCI cards for your Asterisk Install**
  - Analog Telephony Devices – Connect to traditional telephony lines and phones
  - Digital Telephony Devices – Selectable E1/T1/J1 PRI ideal for creating high density solutions
  - BRI –European Basic Rate Interface Card
  - Voice Processing – G.729/G.723 Transcoding card



# Asterisk Add-Ons



- G.729 - Reduces the network bandwidth used by each VoIP call, without sacrificing call quality
- Digium Phone Module for Asterisk - The DPMA is a binary Asterisk module that provides a secure communications channel between Digium phones and Asterisk which manage the phones and provides direct access to Asterisk's internal applications.
- High Performance Echo Cancellation - Digium's High Performance Echo Cancellation (HPEC) module can help eliminate the most common types of echo heard on PSTN connections.

# The Series Continues

Please join us for the next webinar in the series covering the Asterisk REST Interface



Thank you!

Q&A



[www.digium.com](http://www.digium.com) • [www.asterisk.org](http://www.asterisk.org)