5 Things You Didn't Know Asterisk Could Do

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About Me

- Co-author of Asterisk: The Future of Telephony with Jim van Meggelen and Jared Smith (http://astbook.asteriskdocs.org)
  - Asterisk: The Definitive Guide coming in March 2011 (http://ofps.oreilly.com)
- Asterisk bug marshal and release manager
- Consultant with more than 6 years of experience specializing in database integration and clustering

Covered in this presentation

- Cool features in Asterisk 1.8
  - LDAP Integration (also in 1.6.2)
  - Google Voice and Google Talk
  - Calendar Integration
  - Distributed Device State with XMPP
  - `PITCH_SHIFT()`

LDAP Integration

- Available in 1.6.2, but not well known
- Working with clients lately to integration LDAP and writing about it
- Allows ability to have a single set of credentials for signing on by your users
- Helps reduce administrative overhead
- Use existing infrastructure to manage your users (OpenLDAP, ActiveDirectory)

Google Voice and Google Talk

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- Accept calls to your Asterisk system via a free web based client that works on multiple platforms (Windows, Linux, OSX) with Google Talk
- Place calls to web clients
- Place calls for free to numbers in the USA and Canada on the PSTN through Google Voice
- Using `JabberSend()`, you can also send XMPP messages for on-screen pop-ups
- Asterisk can accept messages with `JABBER_RECEIVE()` to re-route calls without answering the line

Calendar Integration

- Allows you to hook your Asterisk system to things like Google Calendar, MS Exchange, or Zimbra to get status from a calendar
- Perform routing logic based on your calendars status
- Redirect calls to voicemail automatically when you're listed as in a meeting
- Or redirect to your cell phone when you're out of the office
- Automatically call participants of a conference with some dialplan logic

Distributed Device State with XMPP

- Distributed device state allows multiple Asterisk systems to know the line status of a device remotely (good for queue distribution, line appearances, message waiting indication)
- Can do it with OpenAIS starting in Asterisk 1.6.1
  - Limited: must be on a low latency network like a LAN
- Starting in Asterisk 1.8, we can do it with XMPP
  - Removes the necessity of being on a low latency link
  - Device state for systems in different physical locations

PITCH_SHIFT()

- Dialplan function that allows you to modify the pitch of an audio channel up or down
- Can be modified on the fly with DTMF via the `features.conf`
- Something fun David Vossel did in his spare time
- Can have a good laugh if you modify the audio of participants randomly prior to them joining a conference room
- Grow your company!

LDAP

Installation of OpenLDAP

- Using the instructions located at https://help.ubuntu.com/10.04/serverguide/C/openldap-server.html to get started with the initial schema configuration
- Once installed, use the `contrib/scripts/asterisk.ldif` file to import the schema for realtime support
- `ldapadd -Y EXTERNAL -H ldapi:/// -f asterisk.ldif`
- After importing, we then import our users, or modify the existing users to support the objectClasses we imported (such as AsteriskSIPUser)

Configuring Asterisk for LDAP

- The key files are:
  - `res_ldap.conf`
    - How to connect to the LDAP server
    - Mapping fields in Asterisk to LDAP schema
  - `extconfig.conf`
    - Tells Asterisk where we're getting configuration information for realtime
    - Enables LDAP support for modules supporting realtime (chan_sip, chan_iax, queues, voicemail, etc.)

Configuring Asterisk for LDAP

- res_ldap.conf

```
[_general]
url=ldap://172.16.0.103:389
protocol=3
basedn=dc=shifteight,dc=org
user=cn=admin,dc=shifteight,dc=org
pass=canada

[sip]
name = cn
callerid = AstAccountCallerID
...
additionalFilter=(objectClass=AsteriskSIPUser)
```

Configuring Asterisk for LDAP

- extconfig.conf

```
[settings]
sipusers => ldap,"ou=users,dc=shifteight,dc=org",sip
sippeers => ldap,"ou=users,dc=shifteight,dc=org",sip
```

Calendar Integration

Installation of Calendar Integration

- Requires libical-dev and libneon-dev
  - CentOS:

```
yum –enablerepo=epel libical-devel
neon-devel
```

  - Ubuntu:

```
apt-get install libical-dev libneon-
dev
```

Enabling Calendar Support

- Uses calendar.conf

```
[myGoogleCal]
type=caldav        ; calendar type
url=https://www.google.com/calendar/dav/
user=leif.madsen@gmail.com   ; login name
secret=welcome               ; password
refresh=15        ; how often to update
timeframe=60      ; range of time to get data
```

Triggering Calls with Google Calendar

- We can trigger calls to devices when events are starting
- May be useful for meeting reminders when you've fallen asleep at your desk after a night at AstriCon
- Acts as a wake up call
  - Perhaps a hotel could use this to allow people to schedule calls for their rooms?
- Add following to the `[myGoogleCal]` section

```
channel=SIP/0000FFFF0001
app=Playback
appdata=this-is-yr-wakeup-call
```

Triggering Calls With Google Calendar

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- Uses calendar.conf

```
[myGoogleCal]
type=caldav
url=https://www.google.com/calendar/dav/
user=leif.madsen@gmail.com
secret=welcome
refresh=15
timeframe=60
channel=SIP/0000FFFF0001
app=Playback
appdata=this-is-yr-wakeup-call
```

Advanced Calendar Call Control

- Dialplan example for setting up call between two meeting participants

```
[AutomatedMeetingSetup]
exten => start,1,Verbose(2,Triggering meeting setup for two participants)
   same => n,Set(DeviceToDial=
                 ${FILTER(0-9A-Za-z,${CALENDAR_EVENT(location)})})
   same => n,Dial(SIP/${DeviceToDial},30)
   same => n,Hangup()
```

Configuring calendar.conf

- We modify `calendar.conf` to execute dialplan upon answer instead of a specific dialplan application

```
[myGoogleCal]
type=caldav
url=
https://www.google.com/calendar/dav/
user=leif.madsen@gmail.com
secret=welcome
refresh=15
timeframe=60
channel=SIP/0000FFFF0001
context=AutomatedMeetingSetup
extension=start
```

Calling and Placing Participants of a Meeting into a Conference Room

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- Examples in next edition of *Asterisk: The Definitive Guide*
- Clever dialplan using `Local` channels and `Originate()` to simultaneously call participants
- Participants don't hear each other until conference organizer joins (administrator)
- Organizer is last person to join the conference

Google Voice / Talk

Google Voice and Google Talk

- Google Talk
  - Ability to place calls between Google accounts with web interface
  - Web interface is multi-platform and free
    - Other web based VoIP clients tend to use ActiveX (Windows Only)
  - Equivalent of softphone-to-softphone communication (on-net only)
- Google Voice
  - Get a free DID number in the USA to accept calls
  - Place calls for free to the USA and Canada
  - Downside: only available to residents of the USA
    - (You can sign up if you connect using a VPN through a USA based server)

Google Voice and Google Talk

- Asterisk 1.8 has chan_gtalk and res_jabber updated
  - Can accept and place calls using Google Talk and Voice
- Configured via `jabber.conf` (user account) and `gtalk.conf` (context control)
- Requires a Gmail account (*@gmail.com*)
  - Can not use a Google Apps account

Configuring jabber.conf

- jabber.conf

```
[general]
debug=no
autoprune=no
autoregister=yes

[asterisk]
type=client
serverhost=talk.google.com
username=asterisk@gmail.com
secret=<secret password>
port=5222
usetls=yes
usesasl=yes
status=available
statusmessage="Asterisk Consulting
```

Configuring gtalk.conf

- `gtalk.conf`
  - Controls where incoming calls are handled in the dialplan
  - Matches on username (email address) first
  - Matches the 's' extension in configured context second
  - Matches on 's' extension in [default] context third

Configuring gtalk.conf

- gtalk.conf

```
[general]
context=default
bindaddr=0.0.0.0
allowguests=yes

[guest]
disallow=all
allow=ulaw
context=gtalk-incoming
connection=asterisk
```

Configuring the dialplan (incoming)

- extensions.conf

```
[gtalk-incoming]
exten => s,1,Verbose(2,Call from ${CALLERID(all)})
    same => n,Answer()
    same => n,Wait(1)
    same => n,Dial(SIP/leifmadsen_desk_HD,30)
    same => n,Voicemail(100@lmentinc,u)
    same => n,Hangup()
```

Placing Calls to Google Talk Users

- extensions.conf

```
exten => 625,1,Verbose(2,Placing a call to malcolmd)
 same => n,Answer()
 same => n,Dial(Gtalk/asterisk/malcolmd@gmail.com,30)
 same => n,Hangup()
```

- Gtalk/***<jabber.conf [section]>***/*<Gmail Address>*

Placing Calls Via Google Voice

- extensions.conf

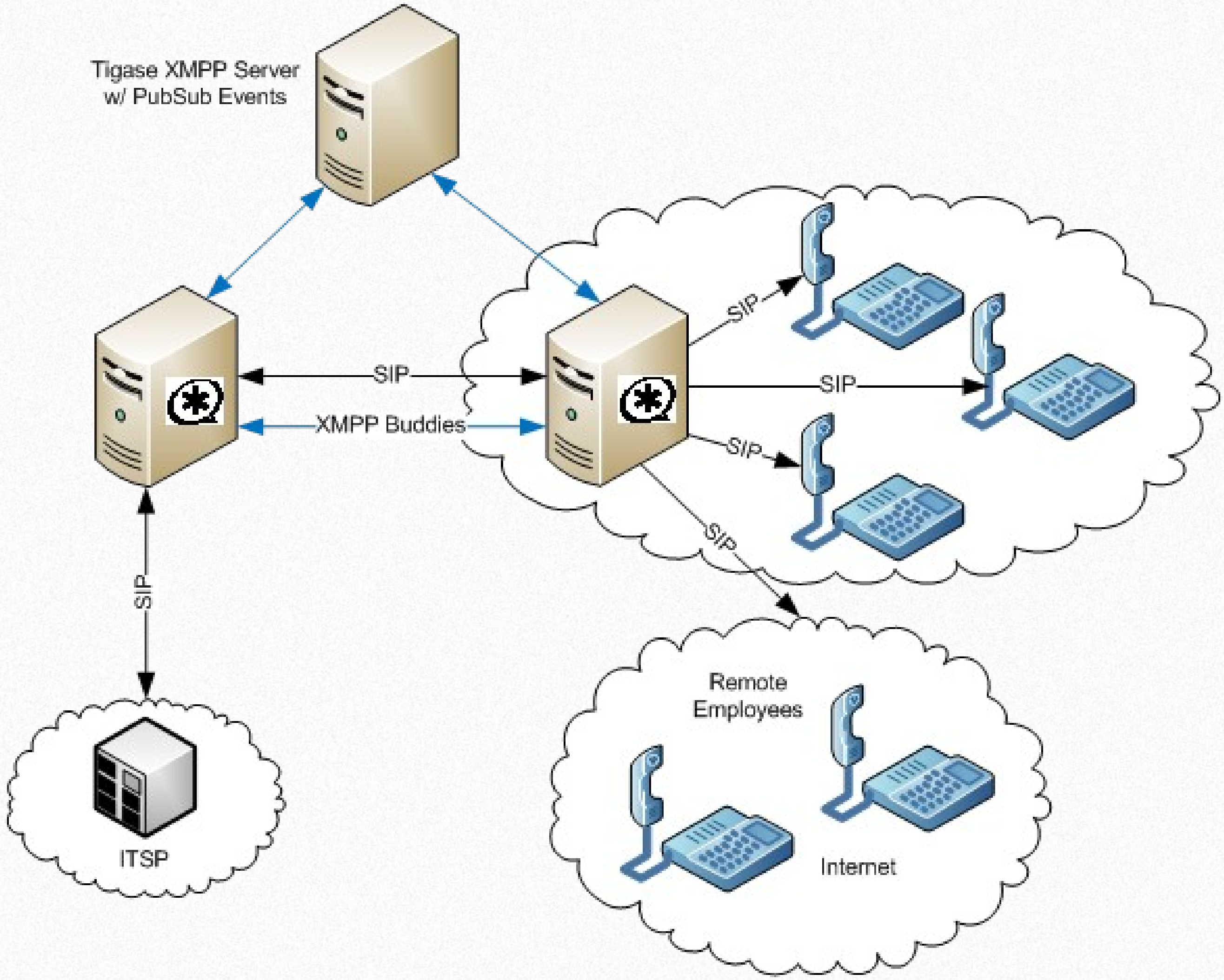
```
exten => _7NXXNXXXXXX,1,Verbose(2,Placing call to ${EXTEN:1})
    same => n,Answer()
    same => n,Wait(1)
    same => n,Dial(Gtalk/asterisk/+1${EXTEN:1}@voice.google.com)
```

Distributed Device State Using XMPP

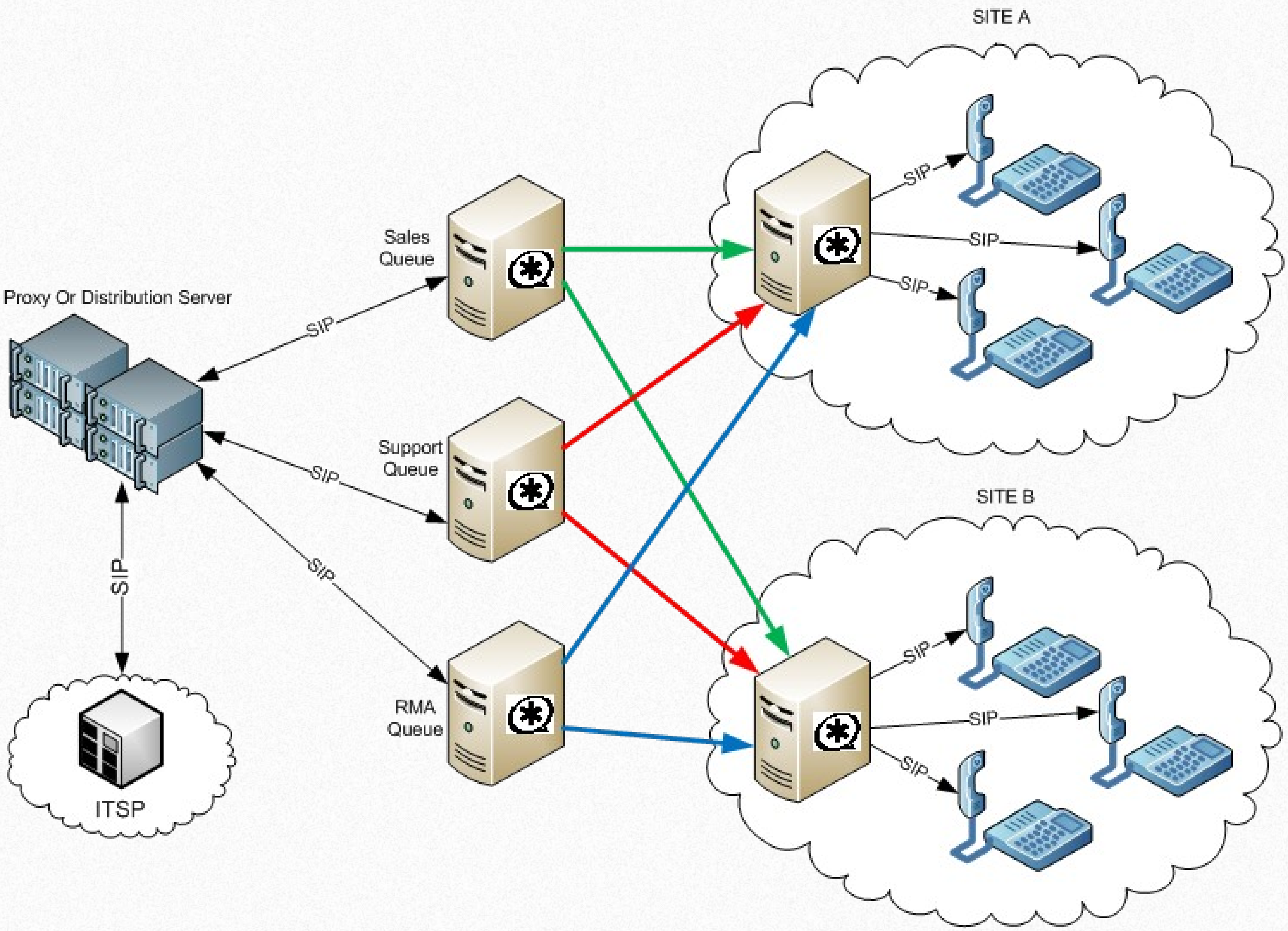
Distributed Device State Using XMPP

- Permits the state of a device (busy, not in use, unavailable) to be distributed across multiple Asterisk servers
- Voicemail MWI as well
- State of devices registered to Asterisk A are known to Asterisk B
- Versions prior to 1.8 have OpenAIS
  - Limited to low latency networks
- Using XMPP allows device state across the internet
- Use Tigase XMPP server (http://www.tigase.org)

Distributed Device State with XMPP



Distributed Device State with XMPP



Configuring Distributed Device State

- Instructions for configuring Tigase server in the `doc/distributed_devstate-XMPP.txt` in Asterisk source
- Uses `jabber.conf` to configure servers that device state is distributed to

Configuring Distributed Device State

- Server #1
  - Users need to be registered using Pidgin

```
[general]
debug=no
;autoprune=yes
autoregister=yes
;collection_nodes=yes
;pubsub_autocreate=yes

[asterisk]
type=client
serverhost=asterisk.mydomain.tld
pubsub_node=pubsub.asterisk.mydomain.tld
username=server1@asterisk.mydomain.tld/astvoip1
secret=welcome
distribute_events=yes
status=available
usetls=no
usesasl=yes
buddy=server2@asterisk.mydomain.tld/astvoip2
```

Configuring Distributed Device State

- Server #2

```
[general]
debug=no
;autoprune=yes
autoregister=yes
;collection_nodes=yes
;pubsub_autocreate=yes

[asterisk]
type=client
serverhost=asterisk.mydomain.tld
pubsub_node=pubsub.asterisk.mydomain.tld
username=server2@asterisk.mydomain.tld/astvoip2
secret=welcome
distribute_events=yes
status=available
usetls=no
usesasl=yes
buddy=server1@asterisk.mydomain.tld/astvoip1
```

PITCH_SHIFT()

PITCH_SHIFT()

- New dialplan application to change pitch of an audio channel
- Designed to be dynamic
  - Can change the pitch of a channel on the fly using features.conf
- Control pitch of audio flowing from party A to B independent of audio flowing from party B to A
- More of a toy for your own entertainment
- What are some examples of business class usage?

Entertaining Uses of PITCH_SHIFT()

- Place calls between two parties at the same time with modified voices and record the result
  - Both parties have the experience of the other side calling them
- Call multiple people at the same time and place them into a conference and randomly adjust the pitch of the users voices
- Randomly assign a pitch change to incoming calls on Fridays for your own entertainment (other side still hears you normally)

Growing Your Company With PITCH_SHIFT()

- Incoming calls enter the auto-attendant and then transfer to your administrative assistant
- Modify the pitch of your voice to the opposite spectrum
- If you don't want to speak to the person, you can say, "*I'm sorry, Mr. Madsen isn't available at the moment, can I transfer you to his voicemail?*"
- Sound like a large company by having different pitches for sales, engineering and your administrative assistant

Questions?

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http://ofps.oreilly.com
(Asterisk: The Definitive Guide, Public Review)

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