Unusual Uses: What you didn't know your Asterisk system could do!

...or, how I learned to love the 1.6 branch

About Me!

- Co-author of Asterisk: The Future of Telephony with *Jim van Meggelen* and *Jared Smith* (http://astbook.asteriskdocs.org)
- Asterisk bug tracker marshal and release manager
- Consultant with more than 5 years experience specializing in database integration and clustering

http://www.leifmadsen.com

Covered in this presentation

- Cool new(ish) features in (or almost in) Asterisk 1.6:
  - IMAP voicemail integration (with greeting storage)
  - New ODBC features (adaptive_cdr_odbc)
  - Calendar Integration
  - CURL
  - XMPP (Jabber) Integration
  - (This presentation based on CentOS 5.x)

IMAP Integration

- Allows you to store your voicemails and emails in the same location
- Repurpose existing IMAP (MS Exchange) infrastructure
- Get to start touting a new (old?) buzzword; Converged!

Adaptive CDR ODBC

- Allows you to store additional call information to the database simply by adding a new column to the database (and writing to it from the dialplan)
- Will automatically create additional columns that the system needs (if the database allows for it)

Calendar Integration

- Allows you to hook your Asterisk system to things like Google Calendar, 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

CURL

- Existed in Asterisk 1.4, but not widely used
- Allows you to get information from a web page and use that information in your dialplan
- Has been used for things like looking up route costs that can be easily managed outside of Asterisk

XMPP Integration

- Can use the XMPP protocol (used by Jabber) to get information to and from Asterisk
- Send a text message from the dialplan to someone
  - Use as a simple way of getting a pop-up on your machine for incoming calls

IMAP

IMAP Integration

- IMAP first appeared in Asterisk 1.4
- Allows us to store voicemail in the same location as our email; Unified Communications! </buzz_word>
- In the 1.6.x branches, we now have the ability to also store greetings in IMAP, and not just on the local file system

Building IMAP Integration

- Need *OpenSSL-devel* and *pam-devel* packages
- On CentOS
  - 64-bit
    - yum install openssl-devel.x86_64 pam-devel.x86_64
  - 32-bit
    - yum install openssl-devel.i386 pam-devel.i386

Building IMAP Integration

- We also need to build the *c-client* libraries from University of Washington
- `wget ftp://ftp.cac.washington.edu/mail/imap.tar.Z`
- Extract it and run:
  - 64-bit
    - `make lr5 EXTRACTFLAGS=-fPIC IP6=4`
  - 32-bit
    - `make lr5 IP6=4`

Install Dovecot

- Then we need to install the IMAP server; Dovecot
- On CentOS
  - 64-bit
    - yum install dovecot.x86_64
  - 32-bit
    - yum install dovecot.i386

Configure Dovecot

- `useradd phonesys`
- `passwd phonesys`
- `mkdir /var/mail/asterisk`
- `mkdir /var/mail/asterisk/phonesys`
- `chown phonesys:phonesys /var/mail/asterisk/phonesys`

Configure Dovecot

- vim /etc/dovecot.conf

```
mail_location = maildir:/var/mail/asterisk/phonesys/%u
protocol imap {

}
auth default {
  mechanisms = plain
  passdb pam {
  }
  passdb passwd-file {
    args = /etc/dovecot.masterusers
    master = yes
  }
  userdb static {
    args = uid=500 gid=500
   }
}
```

Configure Dovecot

- Need to allow Asterisk to authenticate for other users
- `touch /etc/dovecot.masterusers`
- Then add to the file

```
phonesys:{PLAIN}phonesys
```

- Then you can restart the Dovecot service

```
service dovecot restart
```

Configure Asterisk with IMAP Support

- Next we get to compile Asterisk with IMAP support

```
./configure -with-imap=/usr/src/libraries/imap/imap-2007e
```

- Then select the `IMAP_STORAGE` option from *Voicemail Build Options* in menuselect
- Now we can reinstall Asterisk after building

```
make install
```

Configure voicemail.conf

- Next we need to configure our voicemail.conf file to tell Asterisk to connect to the IMAP server

```
imapserver=localhost
imapflags=notls
imapgreetings=yes        ; <-- new!
authuser=phonesys
authpassword=phonesys
expungeonhangup=yes
```

Configure Voicemail Users

- And then in voicemail.conf, we can configure which mailbox our voicemails should be stored in
- We can also use the `imapsecret` option if we needed to authenticate with the server as our peer (not necessary in our case)

```
[imapvoicemail]
100 => 1234,Sue's Mailbox,,,imapuser=sue@example.tld
101 => 5555,Bob's Mailbox,,,imapuser=bob@example.tld
```

Sorry, nothing fancy here :)

- Once you have everything setup and running, your Voicemail() and VoicemailMain() applications just work the same as before!
- (I promise some dialplan and such coming up!)

Adaptive ODBC

Adaptive ODBC

- Allows Asterisk to 'adapt' to table layouts
- Can add columns it expects and needs
- Lets you create new columns, and access them from the dialplan (such as adding a custom value to your CDRs)
- Minimizes the amount of work required to get tables setup for the Asterisk Realtime Architecture (ARA)

Building Adaptive Capabilities

- Need the *unixODBC-devel* and *libtool-ltdl-devel* packages
- On CentOS run
  - 64-bit
    - yum install unixODBC-devel.x86_64 libtool-ltdl-devel.x86_64
  - 32-bit
    - yum install unixODBC-devel.i386 libtool-ltdl-devel.i386

MySQL ODBC

- If you want to use MySQL with ODBC, then you will need to also install the *mysql-connector-odbc* package
  - yum install mysql-connector-odbc
- If you want to use *res_mysql*, then you need to install *asterisk-addons* and the *mysql-devel* development headers
  - yum install mysql-devel

CDR Adaptive ODBC

- The start of the adaptive realtime engine
- Allows you to omit data you don't want to log by not including the column in your table
- Create aliases for column names in cdr_adaptive_odbc.conf
- Now you can adapt Asterisk to your own table layouts!

Going Beyond CDRs

- With the advantages the adaptive engine provided to CDRs, it was taken a step further
- With the ARA (realtime), it would fail previously if you were missing a column
- Now Asterisk will warn you about the missing column, and adapt the SELECT, UPDATE, and INSERT queries to the current table layout

No More Broken Realtime!

- If your table layout wasn't exactly what Asterisk expected, it just wouldn't work
- If the developers wanted to add a new column for a new feature and you updated, that new column would cause your existing realtime install to stop working

Doing The Work For You

- If you use the *res_config_pgsql* or *res_config_mysql* modules, Asterisk can even create the missing columns for you
- res_pgsql.conf (stock) and res_mysql.conf (addons) gives you the *requirements* option

Doing The Work For You

- warn: provide a warning about missing columns, types, or lengths
- createchar: create column as a CHAR with appropriate length
- createclose: create column as appropriate type and length
- On occasion may even widen a column for you (if necessary)

Configuring res_mysql.conf

- This is the file where we define our connection to the database

```
[asterisk]
dbhost = 127.0.0.1
dbname = asterisk
dbuser = asterisk
dbpass = asterisk
dbport = 3306
dbsock = /tmp/mysql.sock
requirements=warn ; or createclose or createchar
```

Configuring Realtime

- Then in extconfig.conf we can configure our SIP registrations (and other realtime things) to store and read data from our MySQL connection

```
[settings]
;iaxusers => odbc,asterisk
;iaxpeers => odbc,asterisk
;sipusers => odbc,asterisk
;sippeers => odbc,asterisk
sipregs => mysql,asterisk
;voicemail => odbc,asterisk
;extensions => odbc,asterisk
;meetme => mysql,conferences
;queues => odbc,asterisk
;queue_members => odbc,asterisk
;musiconhold => mysql,asterisk
;queue_log => mysql,asterisk
```

Getting Warned

```
[Apr  7 23:06:33] ERROR[580]: res_config_mysql.c:226 find_table: Failed to query database columns: Table 'asterisk.sipregs' doesn't exist
[Apr  7 23:06:33] ERROR[580]: res_config_mysql.c:561 update_mysql: Table 'sipregs' does not exist!!
[Apr  7 23:06:43] WARNING[580]: res_config_mysql.c:375 realtime_mysql: MySQL RealTime: Failed to query database: Table 'asterisk.sipregs' doesn't exist
```

- Once we've configured res_mysql.conf then we get warned that we're missing the table to store our SIP registrations into
- Now the administrator knows what is missing (this is new!)

And Then There Were Columns!

- Enable *createclose* in res_mysql.conf, create your table, and start Asterisk

```
mysql> CREATE TABLE sipregs (
    -> id int NOT NULL AUTO_INCREMENT,
    -> PRIMARY KEY (id)
    -> );
Query OK, 0 rows affected (0.00 sec)
```

  - Before...

```
mysql> describe sipregs;
+-------+---------+------+-----+---------+----------------+
| Field | Type    | Null | Key | Default | Extra          |
+-------+---------+------+-----+---------+----------------+
| id    | int(11) | NO   | PRI | NULL    | auto_increment |
+-------+---------+------+-----+---------+----------------+
1 row in set (0.00 sec)
```

...And After!

- Columns automatically created for us!

```
mysql> describe sipregs;
+-------------+----------------------+------+-----+---------+----------------+
| Field       | Type                 | Null | Key | Default | Extra          |
+-------------+----------------------+------+-----+---------+----------------+
| id          | int(11)              | NO   | PRI | NULL    | auto_increment |
| name        | char(10)             | YES  |     | NULL    |                |
| ipaddr      | char(15)             | YES  |     | NULL    |                |
| port        | smallint(5) unsigned | YES  |     | NULL    |                |
| regseconds  | int(10)              | YES  |     | NULL    |                |
| defaultuser | char(10)             | YES  |     | NULL    |                |
| fullcontact | char(35)             | YES  |     | NULL    |                |
| regserver   | char(20)             | YES  |     | NULL    |                |
| useragent   | char(20)             | YES  |     | NULL    |                |
+-------------+----------------------+------+-----+---------+----------------+
9 rows in set (0.00 sec)
```

Calendar Integration

Calendar Integration

- Currently in a branch and can be tracked at http://bugs.digium.com/view.php?id=14771
- Works with MS Exchange, Zimbra, and Google Calendar
- Currently 'Ready for Testing'
- My examples will be with Google Calendar

Calendar Integration

- We can perform call routing decisions based on calendar status; for example, send calls to Voicemail() when you're busy
- Automatically call participants for a conference when you schedule it
- Usage of functions may still change prior to release; additional functions may be necessary

Building Calendar Integration

- Depends on *libical-devel* package from EPEL
- EPEL installation RPM available at http://fedoraproject.org/wiki/EPEL
- On CentOS with EPEL repo installed:
  - 64-bit
    - yum install libical-devel.x86_64
  - 32-bit
    - yum install libical-devel.i386

Configuring for Google Calendar

- Default configuration file calendar.conf contains examples for MS Exchange, Zimbra, and Google Calendar

```
[asterisk-gcal]
type = caldav  ;  type of calendar--currently supported: ical, caldav, or exchange
; Main GMail calendar (the trailing slash is significant!)
url = https://www.google.com/calendar/dav/leif@leifmadsen.com/events/
user = leif@leifmadsen.com ; username
secret = welcome          ; password
refresh = 60               ; refresh calendar every n seconds
timeframe = 60             ; number of minutes of calendar data to pull for each refresh period
                           ; should always be >= refresh / 60
```

Configuring for Google Calendar

- We can show the events for the calendar modules after we reload it

```
freud*CLI> calendar show calendar asterisk-gcal
Name                 : asterisk-gcal
Notify channel       :
Notify context       :
Notify extension     :
Notify applicatio    :
Notify appdata       :
Refresh time         : 60
Timeframe            : 60
Autoreminder         : 0
Events
------
Summary     : Meeting
Description : x100\,p6474483075\,dSIP/leif@leifmadsen.com
Organizer   : mailto:leif@leifmadsen.com
Location    : 6060
UID         : bsdg3e001hp4sghu53vbpauo5o@google.com
Start       : 2009-04-07 03:00:00 PM
End         : 2009-04-07 04:00:00 PM
Alarm       :
```

Routing Calls When Busy

- We can create a simple dialplan that will first check our status to determine if we're busy, and if so, to route calls to Voicemail() instead of ringing our devices

```
  -- Executing [100@phones:1] Verbose("SIP/lmadsen-lmentinc-b409f150", "2,Checking if extension 100 is free") in new stack
 == Checking if extension 100 is free
  -- Executing [100@phones:2] Set("SIP/lmadsen-lmentinc-b409f150", "myCalendarStatus=1") in new stack
  -- Executing [100@phones:3] GotoIf("SIP/lmadsen-lmentinc-b409f150", "1?voicemail") in new stack
  -- Goto (phones,100,6)
  -- Executing [100@phones:6] VoiceMail("SIP/lmadsen-lmentinc-b409f150", "100@lmentinc,b") in new stack
  -- <SIP/lmadsen-lmentinc-b409f150> Playing '/var/spool/asterisk/voicemail/lmentinc/100/busy.slin' (language 'en')
```

Routing Calls When Busy

```
exten => 100,1,Verbose(2,Checking if extension ${EXTEN} is free)
exten => 100,n,Set(myCalendarStatus=${CALENDAR_BUSY(asterisk-gcal)})
exten => 100,n,GotoIf($["${myCalendarStatus}" = "1"]?voicemail)
exten => 100,n,Dial(SIP/lmadsen-lmentinc,30,o)
exten => 100,n,Playback(silence/1)
exten => 100,n(voicemail),Voicemail(100@lmentinc,${IF($["${DIALSTATUS}" = "BUSY" | "${myCalendarStatus}" = "1"]?b:u)})
exten => 100,n,Hangup()
```

- Use the CALENDAR_BUSY() function to get a '1' or '0' when busy, or not busy
- Go right to Voicemail() with busy status if we're not available currently

Automatically Call Meeting Participants

- With some clever tricks, we can automatically call people we want to participate in our conference call – and connect them to the conference room!
- We configure calendar.conf to call a Local channel, then use the Originate() dialplan function.

Automatically Call Meeting Participants

- Remember this part in the Description field? `Description : x100\,p6474483075\,dSIP/leif@leifmadsen.com`
- First character tells us what we're calling:
  - x: local extension
  - d: local device
  - p: phone number

Configure Auto Dial

- In the calendar.conf file we can configure it to connect to the dialplan when it encounters a new busy status
- From there we can get information from the calendar, such as data in the description and location fields
- We use CALENDER_EVENT() for this

```
[calendar]
exten => tryCall,1,Verbose(2,Calendar is looking to call someone)
exten => tryCall,n,Set(DESCRIPTION=${CALENDAR_EVENT(description)})
exten => tryCall,n,Set(CONFERENCE=${CALENDAR_EVENT(location)})
exten => tryCall,n,GotoIf($[${ISNULL(${DESCRIPTION})}]?exit,1)
exten => tryCall,n,Set(AUTOCALL=${CUT(DESCRIPTION,-,1)})
exten => tryCall,n,GotoIf($["${AUTOCALL}" = "AUTOCALL"]?autocall,1:exit,1)

exten => autocall,1,Verbose(2,Attempting to call people in description)
exten => autocall,n,Set(OFFSET=2)
exten => autocall,n,Set(WHO=${CUT(DESCRIPTION,-,${OFFSET})})

exten => autocall,n,While($["${WHO}" != ""])
exten => autocall,n,Set(METHOD=${WHO:0:1})
exten => autocall,n,GotoIf($["${METHOD}" = "x"]?extension)
exten => autocall,n,GotoIf($["${METHOD}" = "p"]?phone)
exten => autocall,n,GotoIf($["${METHOD}" = "d"]?device)
exten => autocall,n,Goto(offset)

exten => autocall,n(extension),NoOp()
exten => autocall,n,Set(EXTENSION=${WHO:1})
exten => autocall,n,Originate(${DB(phones/${EXTENSION}/tech)}/${DB(phones/${EXTENSION}/username)},app,MeetMe,${CONFERENCE}\,d)
exten => autocall,n,Verbose(2,Fall-through)
exten => autocall,n,Goto(offset)

exten => autocall,n(phone),NoOp()
exten => autocall,n,Set(PHONE=${WHO:1})
exten => autocall,n,Originate(${GLOBAL(G_PRIM_ITSP)}/${PHONE},app,MeetMe,${CONFERENCE}\,d)
exten => autocall,n,Goto(offset)

exten => autocall,n(device),NoOp()
exten => autocall,n,Set(DEVICE=${WHO:1})
exten => autocall,n,Originate(${DEVICE},app,MeetMe,${CONFERENCE}\,d)
exten => autocall,n,Goto(offset)

exten => autocall,n(offset),Set(OFFSET=$[${OFFSET} + 1])
exten => autocall,n,Set(WHO=${CUT(DESCRIPTION,-,${OFFSET})})
exten => autocall,n,EndWhile()

exten => autocall,n,Goto(exit,1)

exten => exit,1,Verbose(2,Done with this calendar event)
exten => exit,n,Hangup()
```

Some Kinks...

- Unfortunately the previous dialplan doesn't currently work
- The Originate() application should fall through, but doesn't seem to when used inside a Local channel
- Currently working with a developer to resolve this somehow... such is the life of a tester!

CURL

Using CURL for call rate tracking

- Lookup rate for international / national calling and track cost for each call
- Uses a simple webpage lookup to get the rate for the call
- Allows you to simply update the rate table on the website side, and not have to change anything in Asterisk
- Could be expanded to become a Lease Cost Routing engine

CURL

- I created a PHP script (with some Internet help) to parse and search a CSV file (http://www.leifmadsen.com/presentations/IT360/20080408/curl-example.phps)
- Asterisk then passes the number being dialed to the website
- The CURL() function then retrieves the data and places it into a variable in the dialplan

Building CURL

- To build the 'res_config_curl', 'res_curl', and 'func_curl' functions, you need to install the CURL development libraries for your system
- On CentOS/RHEL:
  - 64-bit
    - yum install curl-devel.x86_64
  - 32-bit
    - yum install curl-devel.i386

Format of CURL()

```
core show function CURL
  -= Info about function 'CURL' =-
[Synopsis]
Retrieves the contents of a URL
[Description]
  url       - URL to retrieve
  post-data - Optional data to send as a POST
(GET is default action)
[Syntax]
CURL(url[,post-data])
```

Setting options for CURL()

Syntax: **CURLOPT(<option>)**

```
cookie        - Send cookie with request
conntimeout   - Number of seconds to wait for connection
dnstimeout    - Number of seconds to wait for DNS response
ftptext       - For FTP, force a text transfer (boolean)
ftptimeout    - For FTP, the server response timeout
header        - Retrieve header information (boolean)
httptimeout   - Number of seconds to wait for HTTP response
maxredirs     - Maximum number of redirects to follow
proxy         - Hostname or IP to use as a proxy
proxytype     - http, socks4, or socks5
proxyport     - port number of the proxy
proxyuserpwd  - A <user>:<pass> to use for authentication
referer       - Referer URL to use for the request
useragent     - UserAgent string to use
userpwd       - A <user>:<pass> to use for authentication
hashcompat    - Result data will be compatible for use with
HASH()
```

Website Output

- URL:
  - http://192.168.128.50/index.php?number=6474483075
- Result:
  - CANADA-647,647,0.011,0.88807702064514,2.1482679843903

Dialplan

```
exten => _NXXNXXXXXX,n,Set(toDial=${EXTEN})
exten => _NXXNXXXXXX,n,Set(RES=${CURL(http://192.168.128.50/index.php?number=${toDial})})
exten => _NXXNXXXXXX,n,Set(ARRAY(country,location,rate,haystack_time,search_time)=${RES})
```

- The above is the “trick” that we're using to get the data from the website, and then writing the values into separate variables

Dialplan

```
exten => _NXXNXXXXXX,1,Verbose(2,CURL Test)
exten => _NXXNXXXXXX,n,Set(toDial=${EXTEN})
exten => _NXXNXXXXXX,n,Set(RES=${CURL(http://192.168.128.50/index.php?number=${toDial})})
exten => _NXXNXXXXXX,n,GotoIf($["${RES}" = "No rate found."]?no_rate,1)
exten => _NXXNXXXXXX,n,Set(ARRAY(country,location,rate,haystack_time,search_time)=${RES})
exten => _NXXNXXXXXX,n,GotoIf($["${rate}" = ""]?no_rate,1)
exten => _NXXNXXXXXX,n,Dial(${GLOBAL(G_PRIM_ITSP)}/${toDial},30)
exten => _NXXNXXXXXX,n,Hangup()

exten => no_rate,1,Verbose(2,No rate found)
exten => no_rate,n,Playback(invalid)
exten => no_rate,n,Hangup()

exten => h,1,Verbose(2,Call cleanup)
exten => h,n,Set(BILLSEC=${CDR(billsec)})
exten => h,n,Set(MINUTES=$[${BILLSEC} / 60])
exten => h,n,ExecIf($["${rate}" != ""]?Set(CALL_COST=$[${MINUTES} * ${rate}]))
exten => h,n,Verbose(2,Cost of this call is ${IF($["${rate}" = ""]?Unknown:${CALL_COST})})
```

Result

```
    -- Executing [86474483075@phones:1] Verbose("SIP/lmadsen-lmentinc-0e172880", "2,CURL Test") in new stack
  == CURL Test
    -- Executing [86474483075@phones:2] Set("SIP/lmadsen-lmentinc-0e172880", "toDial=6474483075") in new stack
    -- Executing [86474483075@phones:3] Set("SIP/lmadsen-lmentinc-0e172880", "RES=CANADA-647,647,0.011,0.81405901908875
,1.0689558982849") in new stack
    -- Executing [86474483075@phones:4] GotoIf("SIP/lmadsen-lmentinc-0e172880", "0?no_rate,1") in new stack
    -- Executing [86474483075@phones:5] Set("SIP/lmadsen-lmentinc-0e172880", "ARRAY(country,location,rate,haystack_time
,search_time)=CANADA-647,647,0.011,0.81405901908875,1.0689558982849") in new stack
    -- Executing [86474483075@phones:6] GotoIf("SIP/lmadsen-lmentinc-0e172880", "0?no_rate,1") in new stack
    -- Executing [86474483075@phones:7] Dial("SIP/lmadsen-lmentinc-0e172880", "SIP/4164790259/6474483075,30") in new st
ack
    -- Called 4164790259/6474483075
    -- SIP/4164790259-0e1679d0 is making progress passing it to SIP/lmadsen-lmentinc-0e172880
    -- SIP/4164790259-0e1679d0 connected line has changed, passing it to SIP/lmadsen-lmentinc-0e172880
    -- SIP/4164790259-0e1679d0 answered SIP/lmadsen-lmentinc-0e172880
    -- Locally bridging SIP/lmadsen-lmentinc-0e172880 and SIP/4164790259-0e1679d0
    -- Executing [h@phones:1] Verbose("SIP/lmadsen-lmentinc-0e172880", "2,Call cleanup") in new stack
  == Call cleanup
    -- Executing [h@phones:2] Set("SIP/lmadsen-lmentinc-0e172880", "BILLSEC=13") in new stack
    -- Executing [h@phones:3] Set("SIP/lmadsen-lmentinc-0e172880", "MINUTES=0.21666666666666667") in new stack
    -- Executing [h@phones:4] ExecIf("SIP/lmadsen-lmentinc-0e172880", "1?Set(CALL_COST=0.0023833333333333334)") in new
 stack
    -- Executing [h@phones:5] Verbose("SIP/lmadsen-lmentinc-0e172880", "2,Cost of this call is 0.0023833333333333334")
 in new stack
  == Cost of this call is 0.0023833333333333334
```

XMPP (Jabber)

XMPP (Jabber) Integration

- Currently have JabberSend() app; first appeared in Asterisk 1.4
- Not widely used; perhaps no one knows about it?
- Branch currently being worked on to give us JabberReceive() (Ready for Testing!)
- http://bugs.digium.com/view.php?id=12569

Building XMPP

- Need to install some dependencies
- On CentOS, need to install EPEL repository
- Depends on *iksemel-devel* and can use *openssl-devel* (for secure connections)
- EPEL installation RPM available at http://fedoraproject.org/wiki/EPEL

Building XMPP

- On CentOS with EPEL repository installed:
  - 64-bit
    - yum install iksemel-devel.x86_64 openssl-devel.x86_64
  - 32-bit
    - yum install iksemel-devel.i386 openssl-devel.i386

Configuring jabber.conf

- It's pretty easy!
- Use your Google talk login, or you can use your company email if using Google apps

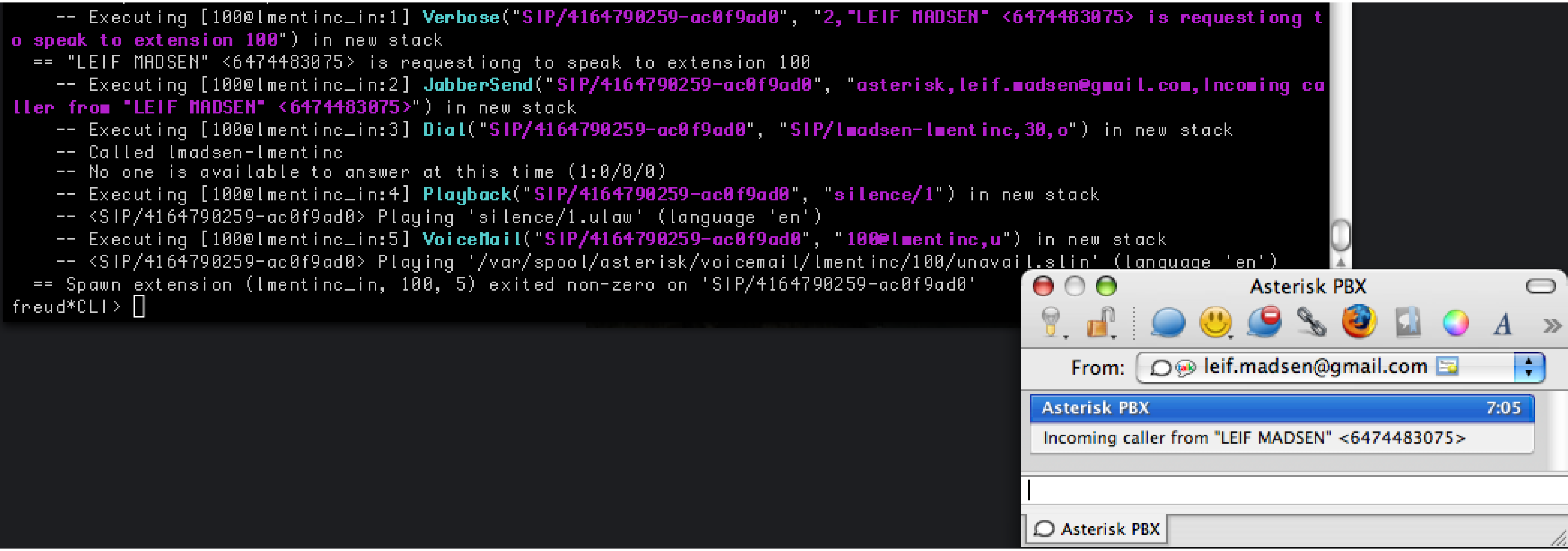
```
[general]
debug=no
autoregister=yes

[asterisk]
type=client
serverhost=talk.google.com
username=asterisk@leifmadsen.com
secret=welcome
priority=1
port=5222
usetls=yes
usesasl=yes
status=available
statusmessage="I am available"
```

Using JabberSend()

- We can create a simple incoming caller pop-up
- Whenever someone calls my extension, an XMPP message pops up to tell me who is calling
- Useful in dark situations because my Polycom IP501 doesn't have a back light

JabberSend() Pop-Up



JABBER_RECEIVE()

- Currently in a branch and ready for testing
- Will go into a future 1.6.x branch (most likely 1.6.3, or potentially 1.6.4)
- Bug tracker location http://bugs.digium.com/view.php?id=12569
- Allows us to receive text from a client and act on it in the dialplan

Call Control via Jabber

- With the JABBER_RECEIVE() function, we can control call flow by sending Asterisk messages
- My example will use JABBER_RECEIVE() and Local channels to control call rejection and forwarding

Call Control via Jabber

- When a call rings my extension, it rings my desk phone, while sending me a message with options

```
Incoming call from "LEIF MADSEN" <6474483075>
Press 1 to send call to voicemail
Press 2 to send call to cell
```

Call Control via Jabber

```
    -- Executing [100@phones:1] Verbose("SIP/lmadsen-cell-ac1481a0", "2,"Leif Madsen" <6474473075> is request
ing to speak to extension 100") in new stack
  == "Leif Madsen" <6474473075> is requesting to speak to extension 100
    -- Executing [100@phones:2] Dial("SIP/lmadsen-cell-ac1481a0", "Local/start@dial-phone&Local/start@receive
-jabber,30,o") in new stack
    -- Called start@dial-phone
    -- Called start@receive-jabber
```

- Call comes into the server and dials extension 100
- Hits the Dial() application and simultaneously calls two contexts via the Local channel

Call Control via Jabber

```
    -- Executing [start@dial-phone:1] Dial("Local/start@dial-phone-cc1c;2", "SIP/lmadsen-lmentinc,,o") in new
 stack
  == Using SIP RTP CoS mark 5
    -- Called lmadsen-lmentinc
    -- Executing [start@receive-jabber:1] Verbose("Local/start@receive-jabber-a918;2", "2,Trying to get data
back from Jabber") in new stack
  == Trying to get data back from Jabber
    -- Executing [start@receive-jabber:2] JabberSend("Local/start@receive-jabber-a918;2", "asterisk,leif.mads
en@gmail.com,Incoming caller from "Leif Madsen" <6474473075>") in new stack
    -- Executing [start@receive-jabber:3] JabberSend("Local/start@receive-jabber-a918;2", "asterisk,leif.mads
en@gmail.com,Press 1 to send to Voicemail") in new stack
    -- Executing [start@receive-jabber:4] JabberSend("Local/start@receive-jabber-a918;2", "asterisk,leif.mads
en@gmail.com,Press 2 to send to Cell") in new stack
    -- SIP/lmadsen-lmentinc-117e2010 is ringing
    -- Local/start@dial-phone-cc1c;1 is ringing
    -- Executing [start@receive-jabber:5] Set("Local/start@receive-jabber-a918;2", "RES=2") in new stack
```

- While calling desk phone, we send options to the Jabber client
- We receive option '2' back and set to the RES channel variable

Call Control via Jabber

```
    -- Executing [start@receive-jabber:6] ExecIf("Local/start@receive-jabber-a918;2", "0?Hangup():NoOp()") in
 new stack
    -- Executing [start@receive-jabber:7] Verbose("Local/start@receive-jabber-a918;2", "2,Answering call beca
use we got data back") in new stack
  == Answering call because we got data back
    -- Executing [start@receive-jabber:8] Answer("Local/start@receive-jabber-a918;2", "") in new stack
    -- Local/start@receive-jabber-a918;1 answered SIP/lmadsen-cell-ac1481a0
  == Spawn extension (dial-phone, start, 1) exited non-zero on 'Local/start@dial-phone-cc1c;2'
    -- Executing [start@receive-jabber:9] GotoIf("Local/start@receive-jabber-a918;2", "0?voicemail,1") in new
 stack
    -- Executing [start@receive-jabber:10] GotoIf("Local/start@receive-jabber-a918;2", "1?cell,1") in new sta
ck
    -- Goto (receive-jabber,cell,1)
    -- Executing [cell@receive-jabber:1] NoOp("Local/start@receive-jabber-a918;2", "") in new stack
    -- Executing [cell@receive-jabber:2] Dial("Local/start@receive-jabber-a918;2", "SIP/4164790259/6474483075
") in new stack
  == Using SIP RTP CoS mark 5
    -- Called 4164790259/6474483075
```

- Since option '2' is send call to cell, we do a Goto() and call out the provider to a cell phone

Making it all work

```
exten => 100,1,Verbose(2,${CALLERID(all)} is requesting to speak to extension ${EXTEN})
exten => 100,n,Dial(Local/start@dial-phone&Local/start@receive-jabber,30,o)
exten => 100,n,Playback(silence/1)
exten => 100,n,Voicemail(100@lmentinc,u)
exten => 100,n,Hangup()
```

- Caller dials extension 100, which calls two local extensions via Local channels
- If we come back with no Answer() after 30 seconds, we fall over to Voicemail()

Making it all work

```
[dial-phone]
exten => start,1,Dial(SIP/lmadsen-lmentinc,,o)

[receive-jabber]
exten => start,1,Verbose(2,Trying to get data back from Jabber)
exten => start,n,JabberSend(asterisk,leif.madsen@gmail.com,Incoming caller from ${CALLERID(all)})
exten => start,n,JabberSend(asterisk,leif.madsen@gmail.com,Press 1 to send to Voicemail)
exten => start,n,JabberSend(asterisk,leif.madsen@gmail.com,Press 2 to send to Cell)
exten => start,n,Set(RES=${JABBER_RECEIVE(asterisk,leif.madsen@gmail.com,20)})
exten => start,n,ExecIf($[${ISNULL(${RES})}]?Hangup():NoOp())
exten => start,n,Verbose(2,Answering call because we got data back)
exten => start,n,Answer()
exten => start,n,GotoIf($["${RES}" = "1"]?voicemail,1)
exten => start,n,GotoIf($["${RES}" = "2"]?cell,1)
exten => start,n,Goto(voicemail,1)

exten => voicemail,1,NoOp()
exten => voicemail,n,Playback(silence/1)
exten => voicemail,n,Voicemail(100@lmentinc,b)
exten => voicemail,n,Hangup()

exten => cell,1,NoOp()
exten => cell,n,Dial(SIP/4164790259/6474483075)
exten => cell,n,Hangup()
```

Webliography

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Webliography

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Contact Information

Leif Madsen

http://www.leifmadsen.com

twitter: leif_madsen

email: leif@leifmadsen.com