Asterisk 14 - Under the Hood

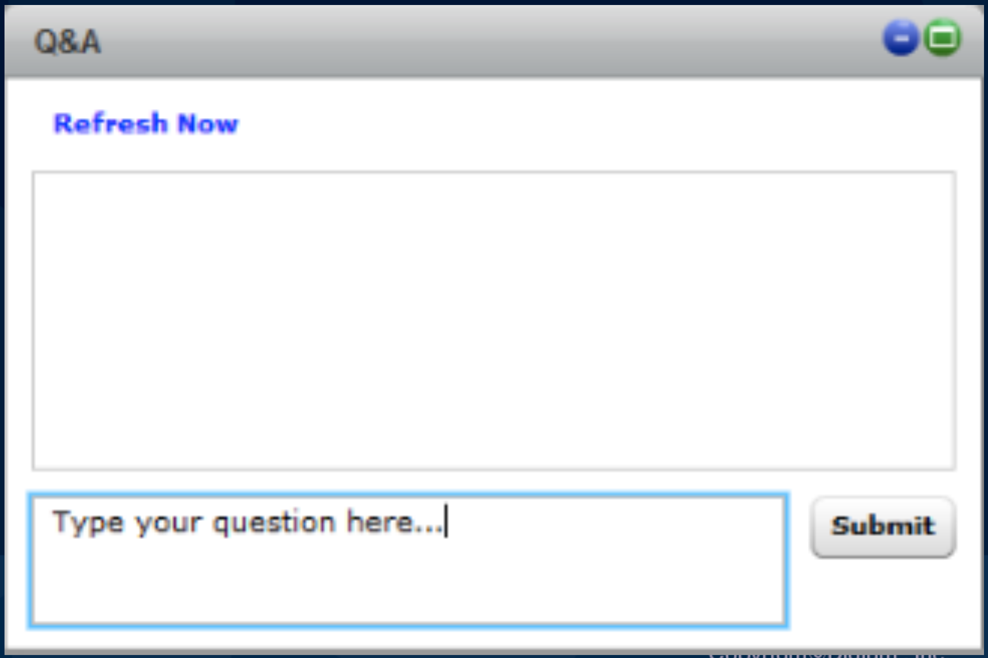
Matthew Fredrickson –Asterisk Project Manager at Digium

@creslin287

Pete Engler – Product Marketing Manager

Housekeeping

- Thank you for attending!
- All attendees are on listen-only mode
- If you have a question, please enter it into the webinar interface
- All questions will be answered at the end of the presentation



Asterisk 14

Personal Background

Who are you and what have you done with Matt Jordan!!?

- Worked at Digium since 2001 in various developmental capacities
- Worked on Asterisk at different times
- Maintained libpri and DAHDI for many years
- Wrote an SS7 stack for Asterisk (libss7)
- Worked on WebRTC related initiatives for the last few years
- Manage the Asterisk project

What’s new with Digium and Asterisk

Matt Jordan moved into CTO position

Hired one of the prominent community contributors, George Joseph

Recently released Asterisk 14

Contribution Statistics for 14

Asterisk 14 contribution statistics:

- 1311 Commits

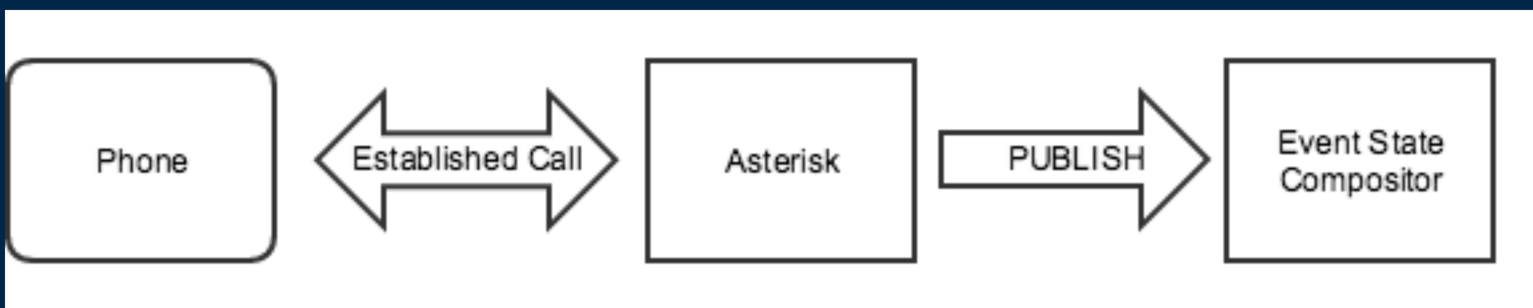
- 91 Individual contributors (according to authorship)

What is new in Asterisk 14? (and some added in 13)

- Asterisk as better infrastructure
- Expand Asterisk’s REST interface
- Make Asterisk easier to get started for new users
- More features

Asterisk as better infrastructure

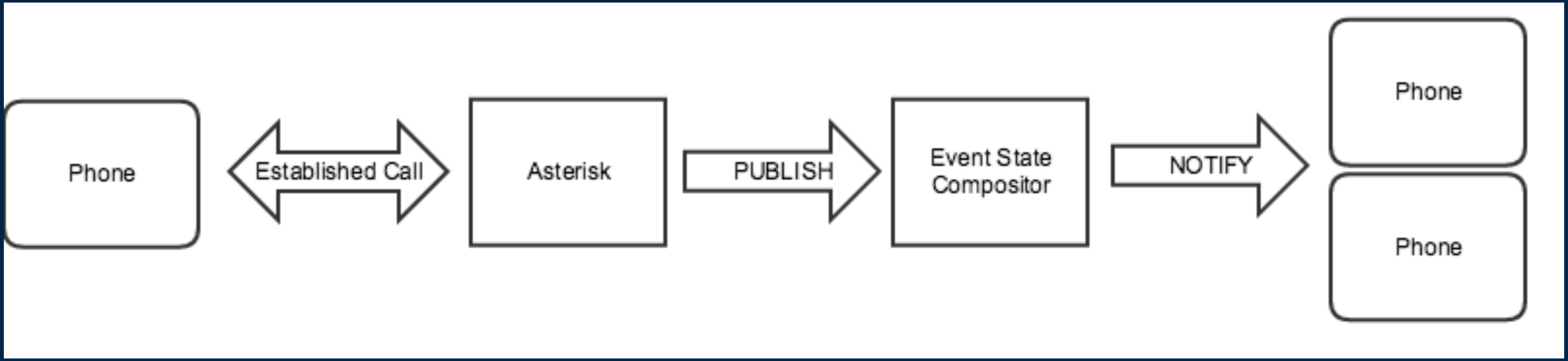
Publishing Extension State - What is it?:



Asterisk as better infrastructure

digium
The Asterisk Company

Publishing Extension State - Why?:



Asterisk as better infrastructure

digium
The Asterisk Company

Publishing Extension State - How?:

In pjsip.conf

```
[test-esc]
type=outbound-publish
server_uri=sip:172.16.0.100
from_uri=sip:172.16.0.100
event=dialog
multi_user=yes
@body=application/dialog-info+xml
@context=^users
```

Asterisk as better infrastructure

Publishing Extension State - The other entity:

Kamailio + presence module

OpenSIPS + presence module

Asterisk as better infrastructure

digium
The Asterisk Company

New DNS API:

- Backed by plugin based resolvers:

- System resolver: supports default system installed DNS resolver

- libunbound resolver - also supports DNSSEC and other cool features

Asterisk as better infrastructure

Goals for new DNS API:

- chan_pjsip now supports NAPTR records

- NAPTR record provide a mapping of services to servers that can handle them for a domain (with order and preference)

- NAPTR records can also specify protocols (TCP vs UDP) and other detailed specifics.

Asterisk as better infrastructure

Goals for new DNS API:

- chan_pjsip now looks for SRV records

- This allows Asterisk to support SRV based load balancing from SIP providers

- chan_pjsip now looks for AAAA records if IPv6 is configured for a transport

Asterisk as better infrastructure

digium
The Asterisk Company

Example scenarios:

Failover:

- When a SIP request is sent but doesn’t receive a response, next target in the transport list is attempted

Load balancing:

- Resolution respects the SRV record priority and weighting

- Example: A SRV record with weight 1 and another with weight 2 - second record would receive 2/3 of the traffic.

Asterisk as better infrastructure

JSON logging support (13 also):

- Allows Asterisk output logging in JSON format instead of its traditional format

To enable:

full => [json]debug,verbose,notice,warning,error

Asterisk as better infrastructure

JSON logging support (13 also) cont’d:

Produces log lines like this:

{"hostname":"mjordan-laptop","timestamp":"2015-09-27 21:44:36","identifiers":{"lwp":14414,"callid":""},"logmsg":{"location":{"filename":"file.c","function":"ast_format_def_unregister","line":181},"level":"VERBOSE","message":"Unregistered format sln32\n"}}

Expand Asterisk's RESTful interface

Ability to retrieve recordings via ARI

- No need to setup FTP server

- Removes need for distributed filesystem

- Makes ARI interface more complete

Expand Asterisk's RESTful interface

digium
The Asterisk Company

URI Media Playback:

- Ability to playback media from remote http sources

- Removes need to have media stored locally

- Again, less need for a distributed file system

Expand Asterisk's RESTful interface

Media Playlists:

- Playback of multiple files in a queue in ARI was slightly painful.

- ARI is an asynchronous interface

- Have to wait on asynchronous completion events and play next media file

- Many times required a client side state machine to manage

Expand Asterisk's RESTful interface

Pre-dial and early media channel support:

- Ability to 'create' a channel prior to dialing, so that channel variables and other operations can be performed on the channel.

- Allows access to the media on an outbound channel prior to it being answered, as well as bridging the early media to another channel.

Expand Asterisk's RESTful interface

ARI PJSIP Push configuration support:

- Obviates the need to have a lot of PJSIP configuration in realtime database

- Utilizes ARI interface to push PJSIP configuration objects directly into Asterisk

Make Asterisk easier to build

Chan_sip to chan_pjpsip transition:

- chan_pjsip and its requirements (PJPROJECT) made building challenging

- Now have a bundled compilation option of PJPROJECT with Asterisk

When building, run configure as follows:

./configure --with-pjproject-bundled

More features

Connection pooling added to res_odbc:

- Allows previously singular ODBC connection to be less of a bottleneck.

- Requires UnixODBC version at least 2.3.2

To enable, in res_odbc.conf:

```
[asterisk]
enabled => yes
dsn => asterisk ; Note value should match an entry in your odbc.ini file
max_connections => 20
```

More features

Connection pooling added to res_odbc - cont’d:

After enabling and running `odbc show all` at the Asterisk command line, you should see this:

ODBC DSN Settings

Name: general

DSN: asterisk

Last connection attempt: 1969-12-31 18:00:00

Number of active connections: 2 (out of 20)

More features

Improved app_confbridge performance:

- Reduced number of DNS queries done in the mixing thread

- Allows Asterisk to handle larger number of participants, while keeping mixing in a more real time context.

More features

And much much more…

(see CHANGES file, UPGRADE file, and commit history for more)

Project Background

Asterisk 11 (LTS) was released in October of 2012

Asterisk 12 was released in December of 2013

Asterisk 13 (LTS) was released in October of 2014

Asterisk 14 was released September 26th of 2016

LTS versus Standard release

- LTS - Long term support

- LTS releases (11, 13) - bug fixes for 4 years, followed by 1 year of only security fixes.

- Standard (12, 14) - bug fixes for 1 year, followed by 1 year of only security fixes.

Why 2 years between 13 and 14?

- Spent a lot of time on testing infrastructure and debugging during the 13 cycle.

- 13 branch allowed new features to be added (only if tests were added too)

- For many workloads, should be (or should become) one of the most stable releases of Asterisk.

- Enough new features that are only in new branch that it's time to release.

2 yrs between 13 and 14: Solution

- Add another year of bug fix support to the 13 branch

- This should make up for the 2 year gap between 13 and 14.

- 14 is still not an LTS, so 1 year bug fix and 1 year security fixes.

codec_opus

- This is not a drill
- Digium's legal department has cleared a release of an opus codec
- Released along with the 14 version of Asterisk and includes a format module and a format attribute module
- codec_opus itself is a binary codec that reports back to a stats server the high use count
- Downloadable via menuselect selection at build/install time.

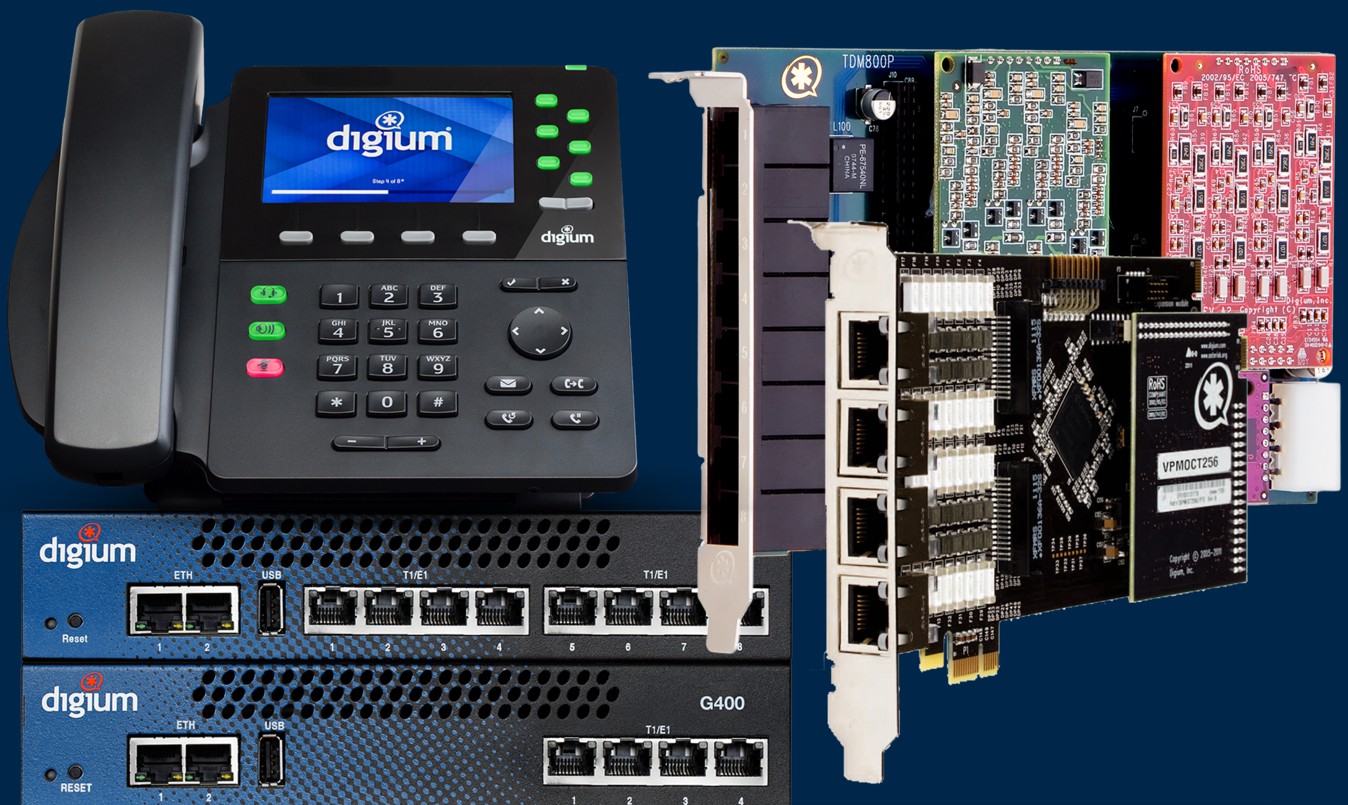
Reminder

- 11 went into security fix only mode in October (get moving forward to 13/14)

Digium Products for Asterisk Solutions

Asterisk Hardware

- Phones
- VoIP Gateways
- Telephony Cards
  - Analog
  - Digital
  - BRI
  - Voice Compression



Digium SIP Trunking

digium
The Asterisk Company

Inbound

- DIDs (US48)
- Toll Free Numbers Available
- Unlimited channels
- Metered or Flat Rate
- CODECs (G711, G729, G722)

Outbound

- Local and Long Distance Metered or Flat Rate
- Free Calls to DCS Subscribers
- G722 (HD voice) Calling Between Subscribers
- E911 DID Registration
- Extended Area and International Calling Available

*SIP Trunking available in the the lower US 48 states only

Thanks!

Matthew Fredrickson

@creslin287 on twitter

creslin@digium.com

Q & A – Please add your questions to the Q&A panel now