Asterisk 7 + 8 = 15

Matthew Fredrickson

@creslin287

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

- Asterisk 14.6.2 and 13.17.2 recently released

- Just released the first full version of Asterisk 15

- Looking forward to what lies ahead with Asterisk

Asterisk 15 contribution statistics:

- 924 Commits

- 82 Individual contributors (according to commit authorship)

General project:

- Almost 2400 merged code reviews on gerrit (for all branches) since DevCon last year.

Top contributors (by # of commits) outside of Digium:

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104 Sean Bright
42 Corey Farrell
39 Alexander Traud
20 Alexei Gradinari
19 Tzafrir Cohen
15 Torrey Searle
11 Walter Doekes
9 Rodrigo Ramírez Norambuena
9 Badalyan Vyacheslav
6 frahaase
6 Sebastian Gutierrez
6 Michael Kuron
5 Daniel Journo
4 kkm
4 Timo Teräs
4 Martin Tomec
4 Joshua Elson
4 Jean Aunis
4 Aaron An
```

What is new in Asterisk 15?

- Platform Improvements

- Miscellaneous Other Improvements

- Video, WebRTC, and more, Oh My!

Platform Improvements

- **GCC 7 fixes**

- **Build fixes for FreeBSD when missing crypt.h**

- **Build fixes for the Gnu HURD**

- **Added support to build against BIND8**

- **OpenSSL 1.1 support**

- **libsrtp2.1 support**

- **Alembic support for MS-SQL**

- **PJPROJECT bundled support is enabled by default**

Miscellaneous Other Improvements

- New Asterisk sounds release (1.6)

- Google OAuth 2.0 protocol support for XMPP/Motif

- chan_rtp uses ulaw by default now instead of slinear

- Binaural audio support patches for confbridge were merged

- debug_utilties: ast_coredumper

- debug_utlities: ast_loggrabber

Video, WebRTC, and more, Oh My!

- Support for RTCP-MUX

- 'webrtc' endpoint option in res_pjsip.conf

- VP9 passthrough support

- RTP dynamic payload numbers are now truly dynamic
(on a per call basis)

- Extensive work to preserve RTP sequence number gaps/
losses across legs in a call (critical for video, makes audio
better too)

- ICE interface blacklist option added to rtp.conf

- Support for more than 32 dynamic RTP payloads now exists.

- Abstracted SDP layer was added (and is still being worked on)

- Added support within the Asterisk core for multi-audio and multi-video stream media per ast_channel

- Added support within the Asterisk core to renegotiate media capabilities on an active call as required

- Support for BUNDLE was added

- app_stream_echo added

- SFU support in app_confbridge

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 Monday, September 26th of 2016

Asterisk 15 was released Tuesday, October 3rd of 2017

- 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.

Asterisk 15 won't be an LTS - but 16 should be.

Due to a lot of the new work that went into master for 15 and much of what's to come (SDP API, ARI improvements for SFU work) it was decided that 15 would not make a great branch for an LTS.

- Chrome decided to require an additional flag be passed in to interoperate with legacy endpoints that lack support for RTCP-MUX in January/February of this year

- Dan Jenkins informed the Asterisk project of this issue around that time

- RTCP-MUX support was implemented at around that time frame to deal with a potential end of life of that behavior

- RTCP-MUX support was merged into Asterisk's 13 and 14 branches

- Chrome is supposed to completely remove support for RTCP-MUX at sometime around the October timeframe

- 11 was already in security fix only mode and is going to be completely dead in October. Get off that branch! (particularly if you run WebRTC)

Thanks!

THANK YOU!