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Asterisk Update

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Asterisk - Year in Review

Asterisk 17.x - 11 bug-fix releases - current is 17.8.0

Asterisk 16.x - 10 bug-fix releases - current is 16.14.0

Asterisk 13.x - 9 bug-fix releases - current is 13.37.0

1750 merged code reviews (across all branches)

Data as of past year as of September 30, 2020

Asterisk - Forum Community Interaction

Accessible at community.asterisk.org

Over 9100 new posts

Over 750 new contributors


drum roll
please...

Asterisk 18 has been released!

421 reviews

56 individual contributors

Asterisk 18 - Top Contributors (by reviews)

71 Joshua C. Colp

68 Sean Bright

58 George Joseph

38 Kevin Harwell

33 Alexander Traud

16 Ben Ford

16 Jaco Kroon

12 Walter Doekes

10 Torrey Searle

9 sungtae kim

8 Corey Farrell

7 Richard Mudgett

6 Guido Falsi

6 Pirmin Walthert

6 Frederic LE FOLL

5 lvl

4 Alexei Gradinari

3 Jean Aunis

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Standard Release vs LTS

Standard Releases:

1 year bug fixes

1 additional year of security fixes

Examples: 17.x

LTS:

4 years of bug fixes

1 additional year of security fixes

Examples: 16.x, 18.x

Asterisk 18 - What's New?

- STIR/SHAKEN
- Video Improvements
- Codec Negotiation Foundation
- Debugging and Logging Improvements
- Miscellaneous Fun!

- Some of these were backported to earlier versions

Asterisk 18 - STIR/SHAKEN

- Standard for attesting permission to use a caller id.
- May be done upstream at provider.
- How it works in practice still under development.
- Within Asterisk we added support for those who may need it.

Asterisk 18 - STIR/SHAKEN

res_stir_shaken:

- Handles configuration for certificates and numbers
- Provides API for signing and verifying
- Provides dialplan functions for examining verification
- Channel driver agnostic (PJSIP supported at the moment)

Asterisk 18 - STIR/SHAKEN

res_pjsip_stir_shaken:

- Interfaces between res_stir_shaken and PJSIP to provide SIP functionality
- Implements SIP specific details of STIR/SHAKEN
- Enabled on a per-endpoint basis
- res_stir_shaken automatically takes care of the rest

Asterisk 18 - STIR/SHAKEN

Dialplan:

- Dialplan function provides inspection of STIR/SHAKEN verification result
- Up to you, based on your own policy, to use the knowledge
- We give you the tools, up to you to use them
- Will be available in Asterisk 16 shortly

Asterisk 18 - Video

Multiple video sources:

- Additional video streams from endpoint are negotiated by PJSIP
- bridge_softmix determines changes and adds/removes additional video streams on other participants as needed
- Allows both camera and screen share at the same time

Asterisk 18 - Video

Packet loss improvements:

- More aggressive at sending NACK to receive missed packets quicker, reducing video freezes
- Growable buffers for extreme network conditions (with upper limit)

Asterisk 18 - Video

Add/Remove Mid-Call:

- Allows call to start as audio only and then transition to video and back
- Other party is negotiated as appropriate based on incoming video
- Also works for video conference bridges
- Does not have to be bidirectional - only one party can opt to provide their video

Asterisk 18 - Video

Local Channel Support:

- Allows a Local channel to be in between PJSIP channels and have video negotiation work
- Passes through negotiation request and result

Asterisk 18 - Codec Negotiation

Background:

- Grown organically over time since the beginning of Asterisk
- Been minimally touched or manipulated
- A key foundation to Asterisk, so understanding it is essential but not really documented
- Taken time to understand it better and determine ways to improve it as people have asked for in the past

Asterisk 18 - Codec Negotiation

Configuration:

- Traditionally there has been little control over codec negotiation
- Configuration options now exist to control codec preference order (Asterisk vs endpoint)
- Additional options exist for future functionality (such as selectively disabling transcoding)

Asterisk 18 - Codec Negotiation

Core:

- No ability to convey negotiated codecs on an outgoing answered call to an incoming leg
- Foundation now in place to allow this to happen
- Will allow the outbound negotiated codecs to be taken into account (based on configuration) on incoming leg
- Support in PJSIP coming in future (wanted to focus on foundation first)

Asterisk 18 - Helping When Things Go Awry

res_pjsip_logger:

- Created to be equivalent to logging functionality in chan_sip
- Hard to examine SIP traces or filter
- Gets the job done, but not the best

Asterisk 18 - Helping When Things Go Awry

res_pjsip_logger:

- Now supports logging to pcap files
- No external dependencies, functionality is always present
- Now also supports specifying multiple IP addresses or subnet masks to filter logging
- All packets are logged in unencrypted (even if they are received or sent encrypted)
- In the future RTP support may also be added

Asterisk 18 - Miscellaneous Fun

- ARI channel create now accepts dialplan variables
- Video stream states now properly conveyed (sendonly / recvonly)
- Optimizations for relaying video to prevent needless thread wakeups
- Improved ICE handling, media will now be sent early if at least 1 valid pair is found
- GCC 10 came into existence and we fixed issues it found

Asterisk 18 - Miscellaneous Fun

- Content type can now be specified to the AMI SendText action
- ARI now allows preventing connected line updates when adding a channel to a bridge
- H.265/HEVC is now a supported video codec
- ARI and AMI events have been reduced by disabling the creation of events for the Message queue channel used for text messaging
- A maximum sample rate can now be set on conference bridges
- Text messaging support can now be disabled on conference bridge participants

Asterisk 18 - Miscellaneous Fun

- Bundled PJPROJECT updated to v2.10
- MOH passthrough support can now be toggled in the dialplan for PJSIP using the PJSIP_MOH_PASSTHROUGH dialplan function
- PJSIP "identify" types can now optionally match on IP address AND port, not just IP address
- The res_musiconhold module now has support for a playlist mode
- RTP blacklist support for ICE addresses has been extended to follow ACL conventions giving greater control

Reminder

- Asterisk 13 is now in security fix only status (6 years - may she rest in peace).
- Asterisk 16 is LTS and goes security fix only in 2022.
- Asterisk 17 is now also in security fix only status.

Final bug fix releases for 13 and 17 will be happening shortly.

Reminder

- Keep track of what's happening in newer (non-LTS) major releases of Asterisk - if you don't, you potentially can experience big surprises when you move forward.
- chan_sip continues its march to the end, so move to chan_pjsip when possible.
- As of Asterisk 19 chan_sip will not be built by default.
- Keep apprised of the Asterisk Versions wiki page

THANK YOU! Follow me @creslin287 on Twitter