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)

<table>
<thead>
<tr>
<th>Rank</th>
<th>Name</th>
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<tbody>
<tr>
<td>71</td>
<td>Joshua C. Colp</td>
</tr>
<tr>
<td>68</td>
<td>Sean Bright</td>
</tr>
<tr>
<td>58</td>
<td>George Joseph</td>
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<td>38</td>
<td>Kevin Harwell</td>
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<td>33</td>
<td>Alexander Traud</td>
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<td>16</td>
<td>Ben Ford</td>
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<td>16</td>
<td>Jaco Kroon</td>
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<td>12</td>
<td>Walter Doekes</td>
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<td>10</td>
<td>Torrey Searle</td>
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<td>9</td>
<td>sungtae kim</td>
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<td>8</td>
<td>Corey Farrell</td>
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<td>7</td>
<td>Richard Mudgett</td>
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<td>6</td>
<td>Guido Falsi</td>
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<td>Pirmin Walthert</td>
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<td>Frederic LE FOLL</td>
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<td>5</td>
<td>lvl</td>
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<td>4</td>
<td>Alexei Gradinari</td>
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<td>3</td>
<td>Jean Aunis</td>
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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