Asterisk - Do I see video in the future?

Matthew Fredrickson

@creslin287

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**

A little bit of history (cont'd)

Asterisk 1.0 - First major release, ISDN support, H.323, MGCP, AGI

[many changes later]

Asterisk 1.6 - Wideband audio, SS7 support

[…some time passes…]

A little bit of history (cont'd)

Asterisk 11 - Beginnings of WebRTC support in chan_sip

Asterisk 12 - chan_pjsip

Asterisk 13 - ARI, PJSIP

Asterisk 14 - More ARI, more PJSIP, and Async DNS

A little bit of history (cont'd)

What about Asterisk 15?

The historical problem with video

- Limited clients: commodity SIP video phones and SIP desktop clients have traditionally have been single stream limited.

- Premium clients with rich end user experience were historically large and very expensive (think Cisco telepresence, Tandberg solutions, etc)

- Video MCU (multipoint control unit) was considered the defacto standard for providing a multi-user experience (so mix everybody's video stream into a brady bunch stream)

- Requires lots of CPU power on the MCU mixing box. "Video is expensive, yo?" - file

Asterisk 15 - Video streams for the common man

Asterisk 15 does video better than any prior version of Asterisk:

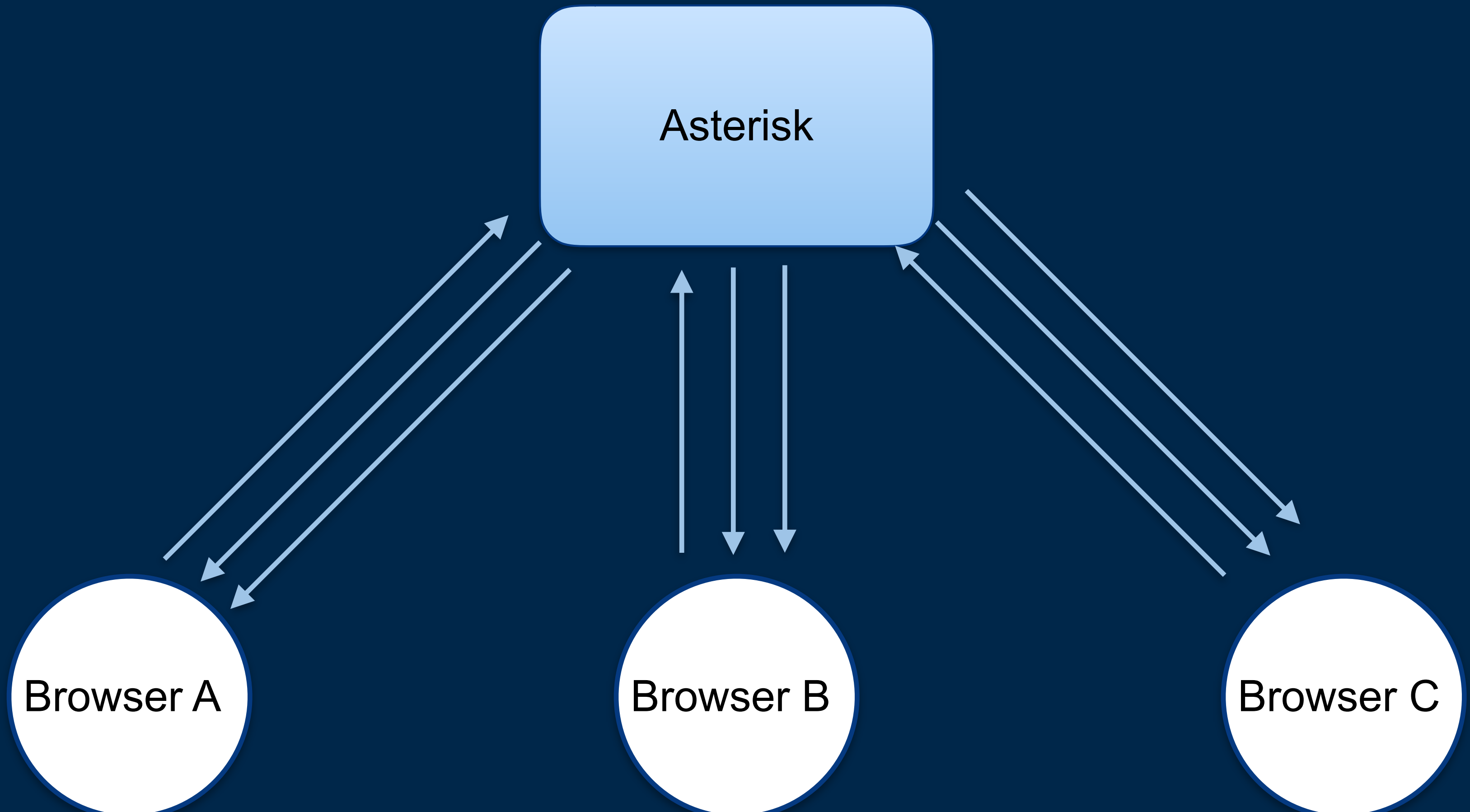
- Multi stream enhancements to the core of Asterisk - the old single-video/single-audio stream per call limitation is broken.

- Asterisk core allows renegotiation of number of video streams and audio streams as well as their attributes on demand.

- app_confbridge now has support to be a generic SFU (selective forwarding unit) - All video streams go to all participants

What is a SFU (Selective Forwarding Unit)?

N participants, each sending one video stream and receiving N-1 video streams from other participants.



Why SFU?

- If one participant wants to have large picture focus on the presenter for a particular period of time, they can.

- If another participant is still taking notes from a screen share, they control their focus (so they can keep the screen share large for an additional period of time if desired)

- Much less CPU usage

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
- 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

- 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
- ICE interface blacklist option added to rtp.conf

Video, WebRTC, and more, Oh My!

- 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

Video, WebRTC, and more, Oh My!

- Support for BUNDLE was added
- app_stream_echo added
- SFU support in app_confbridge

Asterisk 16 - What's next?

- Fleshing out Asterisk's SFU APIs

- Improving Asterisk's video resilience in poor network environments

- PJSIP performance improvement

Contribution Statistics for 15

Asterisk 15 contribution statistics:

- 924 Commits

- 82 Individual contributors (according to commit authorship)

General project statistics:

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

Contribution Statistics for 15

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

```
104 Sean Bright
42 Corey Farrell
39 Alexander Traud
20 Alexei Gradinari
19 Tzafrir Cohen
15 Torrey Searle
11 Walter Doekes
```

Reminder

- 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)

- Asterisk 15 will not be an LTS - but 16 should get us back on track and be the next LTS.

Thanks!

THANK YOU!

digium

The Asterisk Company

Empowering Communication

www.digium.com · www.asterisk.org