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Asterisk - What's Happening in Master?

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Personal Background

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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**

Asterisk-land - What's happening?

- **Recently released version 13.15.0 of the 13 branch of Asterisk and 14.4.0 release of the 14 branch.**
- **Completed certification testing of the 13.13-certified release of Asterisk.**
- **Dennis Guse and Frank Haase's binaural audio patches for Asterisk and app_confbridge have recently been merged.**
- **Jitter buffer improvements to better support features such as FEC in codec_opus**

What's happening in Asterisk's master branch

Quiz: 3 are lies and 3 are true - which channel drivers in Asterisk utilize SDP as a means of conveying media descriptions?

- **chan_sip**
- **chan_jingle**
- **chan_pjsip**
- **chan_mgcp**
- **chan_webrtc**
- **chan_sdp**

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Abstracted SDP layer:

- **Many telecom protocol implement some form of SDP to negotiate media stream attributes: SIP, MGCP, Native browser RTCPeerConnection (JSEP/WebRTC)**
- **Rather than reimplement SDP parsing and abstraction in every channel driver within Asterisk, instead a non-channel driver specific abstract SDP layer should be used.**

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Abstracted SDP layer:

- **Top level user interface API has an offer/answer state and management built in**
- **Handles cases like offer/answer negotiation failure and sdp rollback**
- **Handles early media scenarios (think 183 Session Progress), where initial sdp is not final sdp (like in 200 OK)**
- **Has pluggable bottom end translator API for text parsing/generation (serialization) side of SDP from internal SDP state. Only implementation is currently res_sdp_translator_pjmedia**

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ast_channel and core gaining multi-stream support:

- **Increased interest in a new class of RTC clients. Widely deployed, with more power and vendor neutral capability then ever before - Web browser.**
- **Multistream support (particularly video) is becoming more and more important**
- **Currently, asterisk's ast_channel interface supports only a single stream of each type (audio, video, text)**

What's happening in Asterisk's master branch

ast_channel and core gaining multi-stream support:

- **Extension to ast_channel interface done in a backwards compatible way**
- **All existing channel APIs should remain compatible, defaulting to a single audio/video stream as per history.**
- **Work done though the new stream topology APIs and new ast_stream_topology structure.**
- **Allows stream renegotiation to occur dynamically in Asterisk's core, and can be done on a per application basis.**

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ast_stream and ast_stream_topology:

- **ast_stream_topology can contain one more more ast_stream object.**
- **ast_stream represents a single audio or video stream with an Asterisk channel or ast_channel**
- **An ast_channel can be requested with a certain topology at creation time or a new topology can be requested on the fly (allowing codec renegotiation to occur dynamically)**
- **ast_channel_request_stream_topology_change() to request it at runtime (and subsequent SDP renegotiation)**

What's happening in Asterisk's master branch

ast_stream and ast_stream_topology:

ast_stream_topology - with 4 ast_streams

- ast_stream 1 - audio send/receive
- ast_stream 2 - video send only
- ast_stream 3 - video receive only
- ast_stream 3 - video inactive

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RTCP-MUX support (also in 13 & 14):

- **Required significant changes to res_rtp_asterisk.c**
- **RTCP-MUX is a webrtc technology used to multiplex RTCP and RTP on the same UDP port. (sounds like IAX :-)**
- **No additional encapsulation layer is required to discriminate between RTP and RTCP packets**
- **Certain RTP payload codes are unusable in order for it to work properly**
- **RTCP-MUX support is required in Chrome 57 *****
- **Completed and in all current releases of Asterisk (13.15.0 and 14.0.4)**

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RTCP-MUX: What is it?

Traditional RTP/RTCP

UDP Port N

RTP - Media packets of codec, defined by payload code X

UDP Port N+1

RTCP - Packet loss, RTT, and other data about RTP stream.

What's happening in Asterisk's master branch

RTCP-MUX

UDP Port N

RTP - Media packets of codec, defined by payload code X

RTCP - Packet loss, RTT, and other data about RTP stream.

What's happening in Asterisk's master branch

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Integration of new SDP API and native multistream support in chan_pjsip:

- **To support the new multistream APIs in Asterisk requires channel driver changes.**
- **Existing channel drivers work, even with new APIs (so we didn't break the world, as we'd feared might happen) BUT only with historical support for single audio and video stream.**
- **chan_pjsip will be the first channel driver to support the new multistream APIs as well as the abstracted SDP layer. This work is in progress as we speak.**

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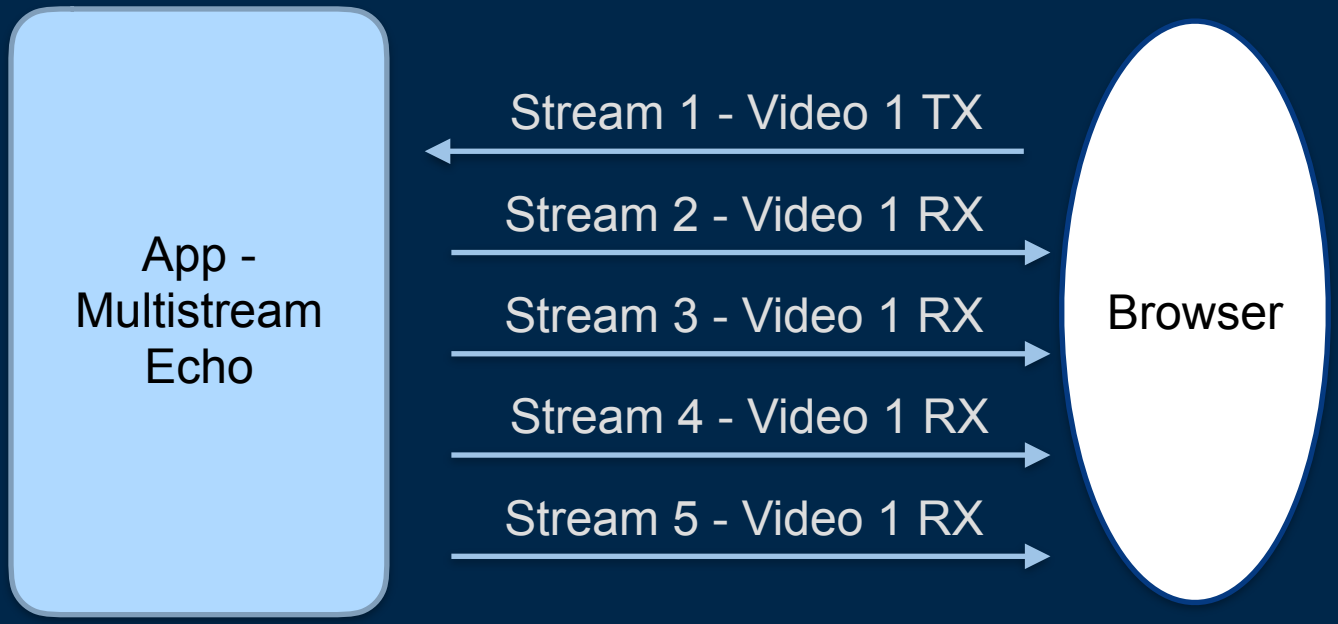
Simple multistream echo application:

- **A lot of work has been done behind the scenes to extend the Asterisk core to work well in a multistream environment.**
- **Many parts have been completed, but some parts are yet in progress (chan_pjsip support, for example)**
- **Need a simple test to do integration testing with a modern browser based endpoint**
- **Multistream echo will open a session to receive one video stream from a browser and echo it back out over N more streams.**
- **This may seem like a simple application, but given that it will be the first end to end integration test of 5 months of code, it's a pretty important.**

What's happening in Asterisk's master branch



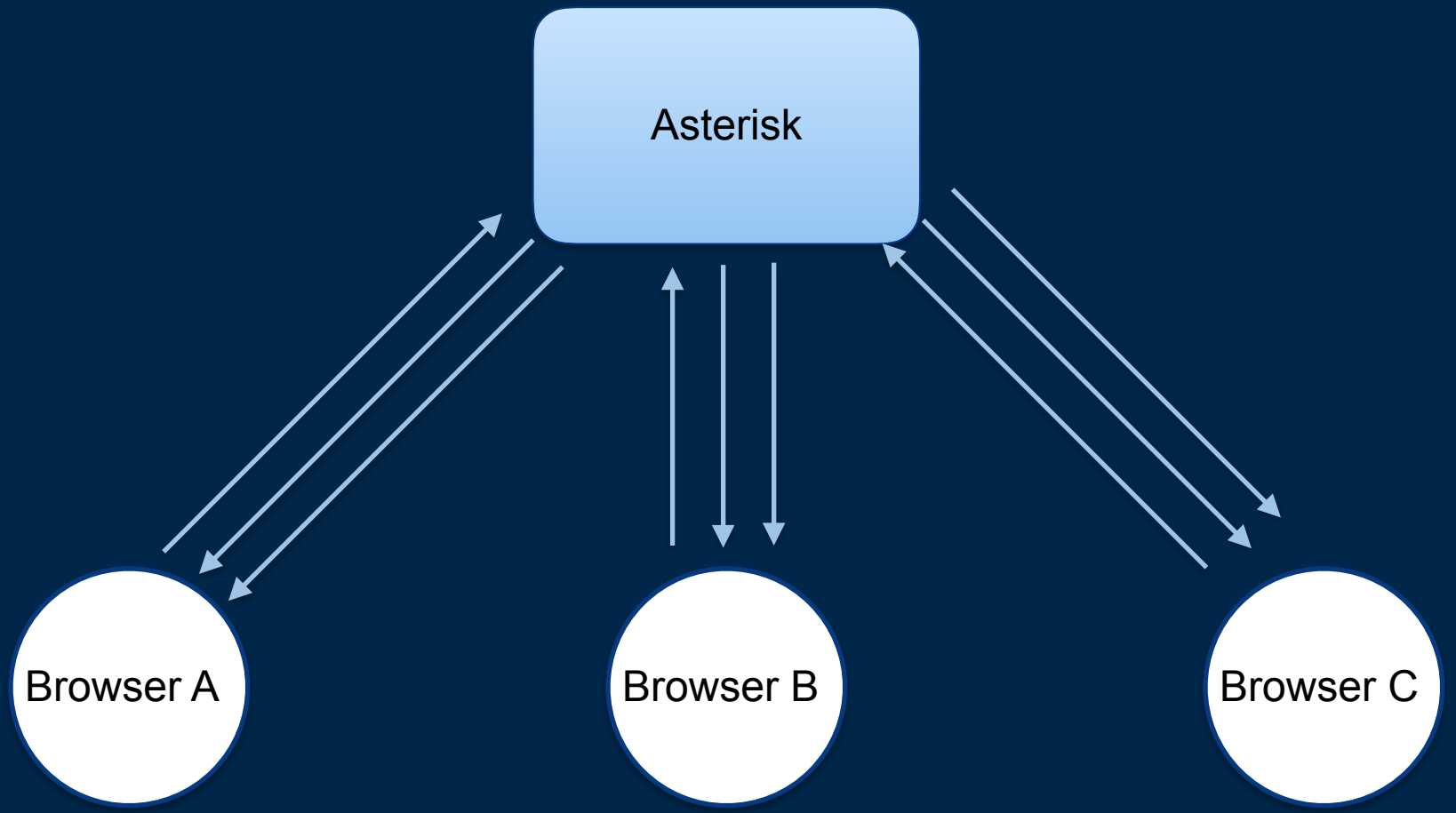
What's happening in Asterisk's master branch

Extend app_confbridge to provide an SFU type experience:

- **For those not familiar, an SFU is a way of forwarding individual video streams from endpoint to endpoint (star network topology) instead of compositing them all together into a single stream like an MCU.**
- **Goal is to have simple, implicit SFU experience in app_confbridge.**

What's happening in Asterisk's master branch

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



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What's next?

- **More steps along the path to fully support the WebRTC media stack**
- **Bundle**
- **Further improvements in app_confbridge SFU support**
- **ARI support for multichannel video setup**
- **ARI hooks for app_confbridge SFU support**

Where is Asterisk going? (next 1-2 years)

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Potential directions:

- **Leave the past: Less SIP. SIP is too new, will never take off, and will never be adopted at any level of significance.**
- **Push to the future: More ISDN - ISDN is the future of radio.**
- **Push to the future: More SS7**
- **Just kidding, of course :-)**

Where is Asterisk going? (next 1-2 years)

Directions:

- **Better handling of flexible, multimedia applications**
- **Improved IoT integration - RTMP channel driver**
- **Continuing to enhance multistream audio and video support within Asterisk's core**
- **Continue to flesh out Asterisk's REST interface (particularly with regards to SFU additions)**

Reminder

- 11 went into security fix only mode in October (get moving forward to 13/14) - it has less than 6 months left to live.

Thanks!

Matthew Fredrickson

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