Asterisk 15
Video Conferencing

The new video conferencing functionality in Asterisk 15
and the journey to get there

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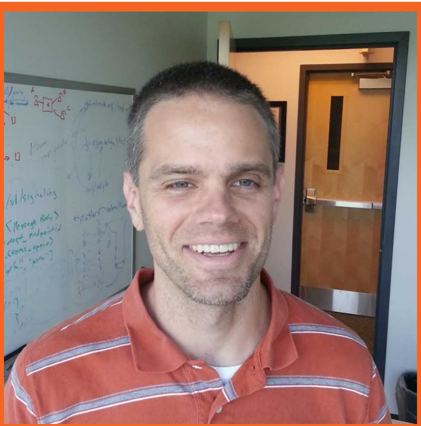

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Overview

- Old Media Flow
- Streams
- New Real Streams
- PJSIP
- Legacy Support
- Bridging
- SFU
- WebRTC

Old Media Flow

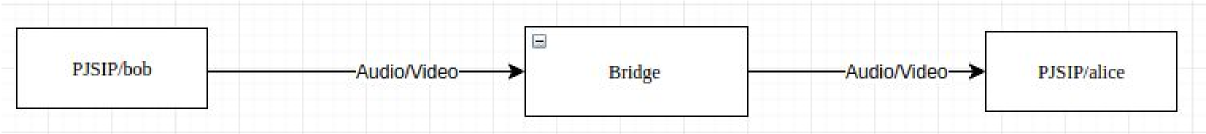
Single logical flow internally carrying media

Each media frame has a type and format

Conceptually only 1 stream of each type possible

Negotiated media formats all combined together

Old Media Flow



Streams

Flow of a single type of media

Can be one way or two way

Has a name which can have meaning

Can be added, removed, or changed

Has negotiated media formats specific to the stream

Requirements for Stream Support

Minimal modifications to Asterisk

Needed to be able to add/remove/change streams

Needed to be simple to use

Needed to be flexible and fast

New Real Streams

First class stream object

Contains only information specific to the stream

Groups of streams are kept in a container called a topology, indexed based on position number

Channel can have streams added, removed, or changed

Each media frame has a type, format, and stream number

Multistream Users

Constructs streams according to negotiated result

Responsible for placing stream topology on channel - not done automatically

Responsible for responding to request for renegotiation

ast_write_stream, ast_read_stream

PJSIP

Only channel driver supporting multiple streams currently

Outgoing uses requested stream topology, adding streams to SDP

Incoming negotiates streams based on configured formats

Can be told to renegotiate to add/remove/change streams

StreamEcho

Extended version of the Echo() application

Will request renegotiation to ensure specified number of streams are present

Echoes media received on first stream of each type to every other stream of that type

Legacy Users

See and interact with only a single pipe like before

Can have only 1 stream of each type

Existing APIs create streams automatically as appropriate

Does not have any knowledge of new stream support

ast_read, ast_write, ast_channel_nativeformats

Required no code changes to legacy users

Legacy Video Support

Calling between devices (if video in initial offer)

Basic video recording

Basic video playback

Conference with single video sent to each participant

Bridging

Currently two bridging modules support multistream:

- Simple (What Dial uses)
- Softmix (What ConfBridge Uses)

Other bridge modules unchanged and behave the same.

How Simple Bridging Now Works

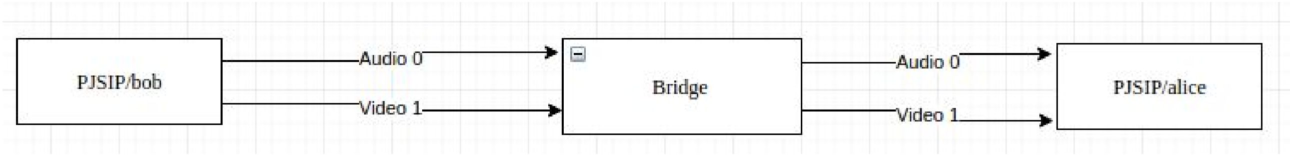
Channel with fewer streams renegotiated to match other

If same number then second channel joined gets renegotiated to match first channel that joined

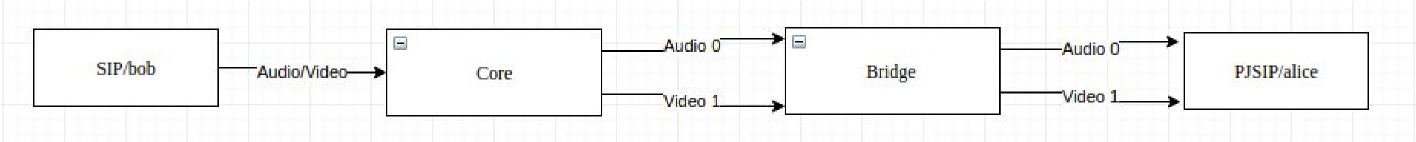
Each stream is mapped 1 to 1

Acts as media forwarder based on stream number

Simple Bridging Flow (Multi)



Simple Bridging Flow (Legacy)



Softmix Additions

Multiple video streams can now be sent to participants (SFU)

SFU

Selective forwarding unit

Picks a subset of video streams to forward

Currently limited by max number of video streams on channel

No server side transcoding or manipulation is done

How Softmix Now Works

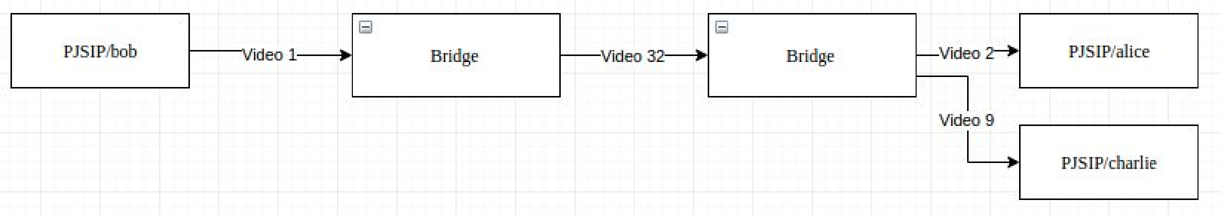
Each video stream on a channel is mapped to a bridge specific stream number

Each channel can have a mapping from bridge specific stream number to channel video stream

Audio is still mixed server side to provide same ConfBridge audio experience as previously

Enabled using video_mode=sfu in ConfBridge

Multiparty SFU Flow



WebRTC

Quick implementation

Best option for rich ConfBridge SFU experience

BUNDLE

Required for Google Chrome to support multiple streams due to Plan B usage

Specification to allow multiple streams to be sent/received over the same transport

Cuts down on ICE and DTLS negotiation time

Now available in PJSIP

CyberMegaPhone

Limited example code available (is on Github, MIT license)

Uses HTML and Javascript

JsSIP based client for use with Asterisk

Adds/removes video as participants join/leave conference

Controls to mute/unmute

Firefox and Chrome supported on desktop

Potential Video Support Additions

Adding/removing video mid-call

Better video recording and playback (with multiple streams)

Feedback allowing video quality to change due to bandwidth change

Better handling of packet loss and out of order packets

Putting It All Together

Streams in Core

Streams in PJSIP

SFU video conferencing in ConfBridge

WebRTC Client

Demo

Questions?

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