Real-Time Applications and Infrastructure Area (RAI)

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Ben Campbell(Presenter) <ben@nostrum.com>
Adam Roach <adam@nostrum.com>
Robert Sparks <rjs@nostrum.com>
Gonzalo Camarillo <gonzalo.camarillo@ericsson.com>
Richard Barnes <rlb@ipv.sx>

What was the area about?

- Tools for letting people interact with each other with minimal delay using the Internet
  - Talking
  - Interactive Video
  - Gaming
  - Live collaborative music
  - Instant Messaging and Presence

Delay Sensitive Interpersonal Communication

RAI Basics

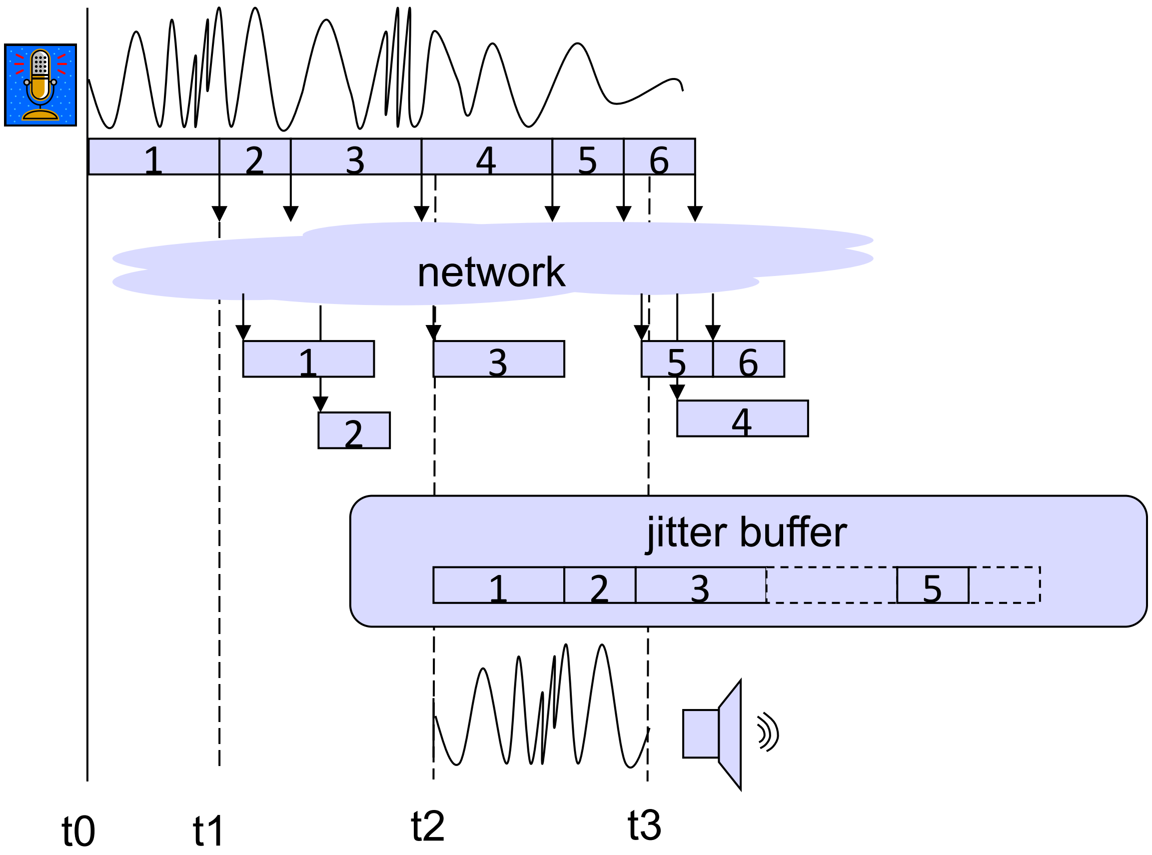
- Building blocks for real-time services
  - Provide (and protecting) location
  - Advertise available real-time services
  - Get emergency calls to the right responder
  - React to a person's changing ability or willingness to communicate

Primary Work

- Move real-time media around (RTP)
- Set up communication sessions (SIP)
- Talk about those sessions (SDP)
- Presence/Messaging (SIMPLE, XMPP)
- Browser Communication (RTCWEB)
- Location, Privacy, and Emergency Services (GEOPRIV, ECRIT)
- Telepresence and Conferencing (CLUE,PERC)

What does RTP do

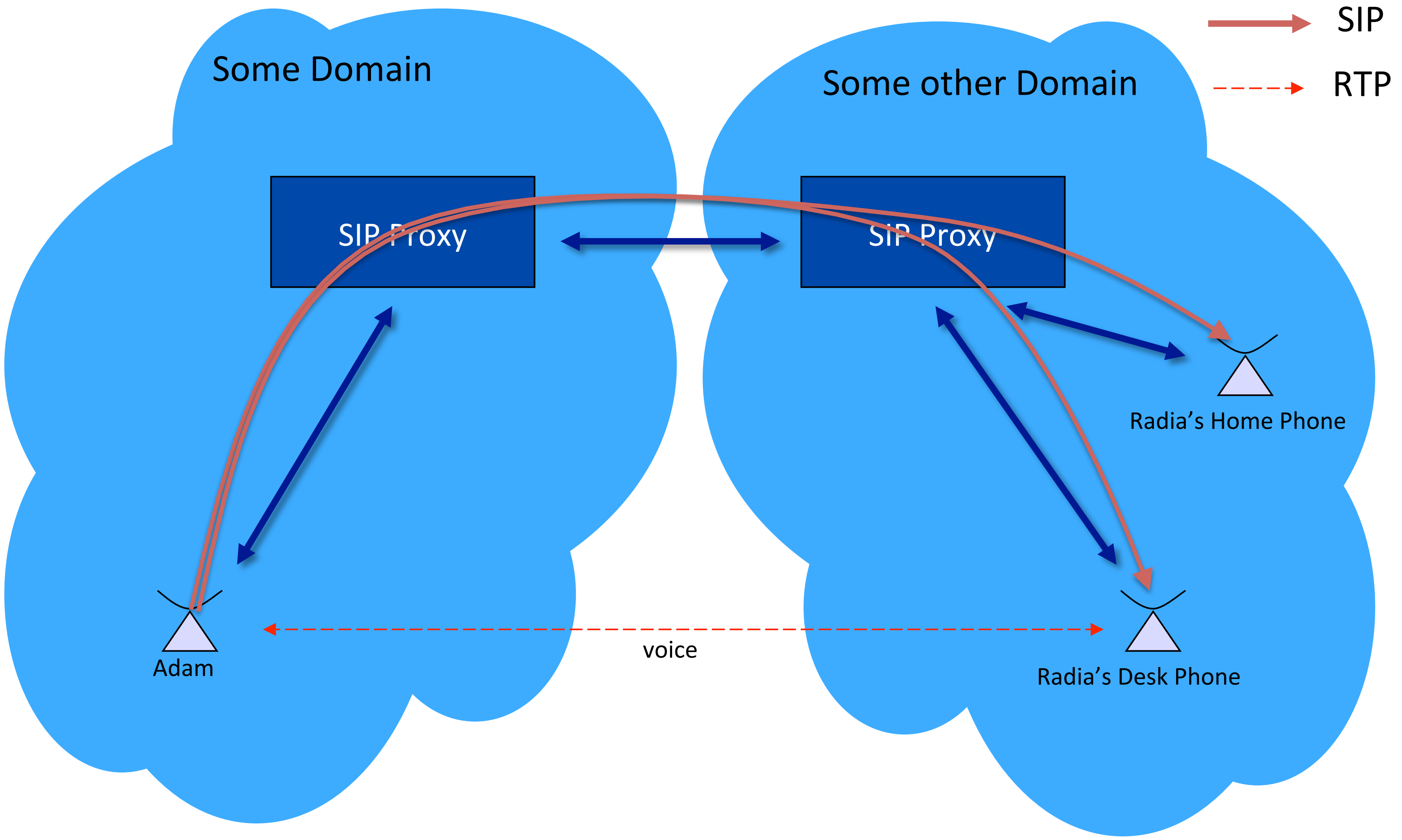
- Carries a time-dependent signal through a packet network, preserving the timing information



What does SIP do?

- Adam wants to talk to Radia. SIP (the Session Initiation Protocol) helps with two things
  - Rendezvous: It helps Adam’s device *find* the right device of Radia’s to work with on the network
  - Negotiation: It lets Adam’s and Radia’s devices determine the technologies they will use to carry the conversation between Adam and Radia.

What does SIP do?



Session Description Protocol (SDP)

- Describes the technologies (and the parameters chosen within those technologies) for communication
- Can be declarative
  - Declaring what a multicast session will contain
  - Used in announcements
- Can be descriptive
  - Describing what an endpoint is willing to do
    - Says things like "I'm willing to receive one audio stream and one video stream".
  - Used in negotiation
    - Offer-Answer model

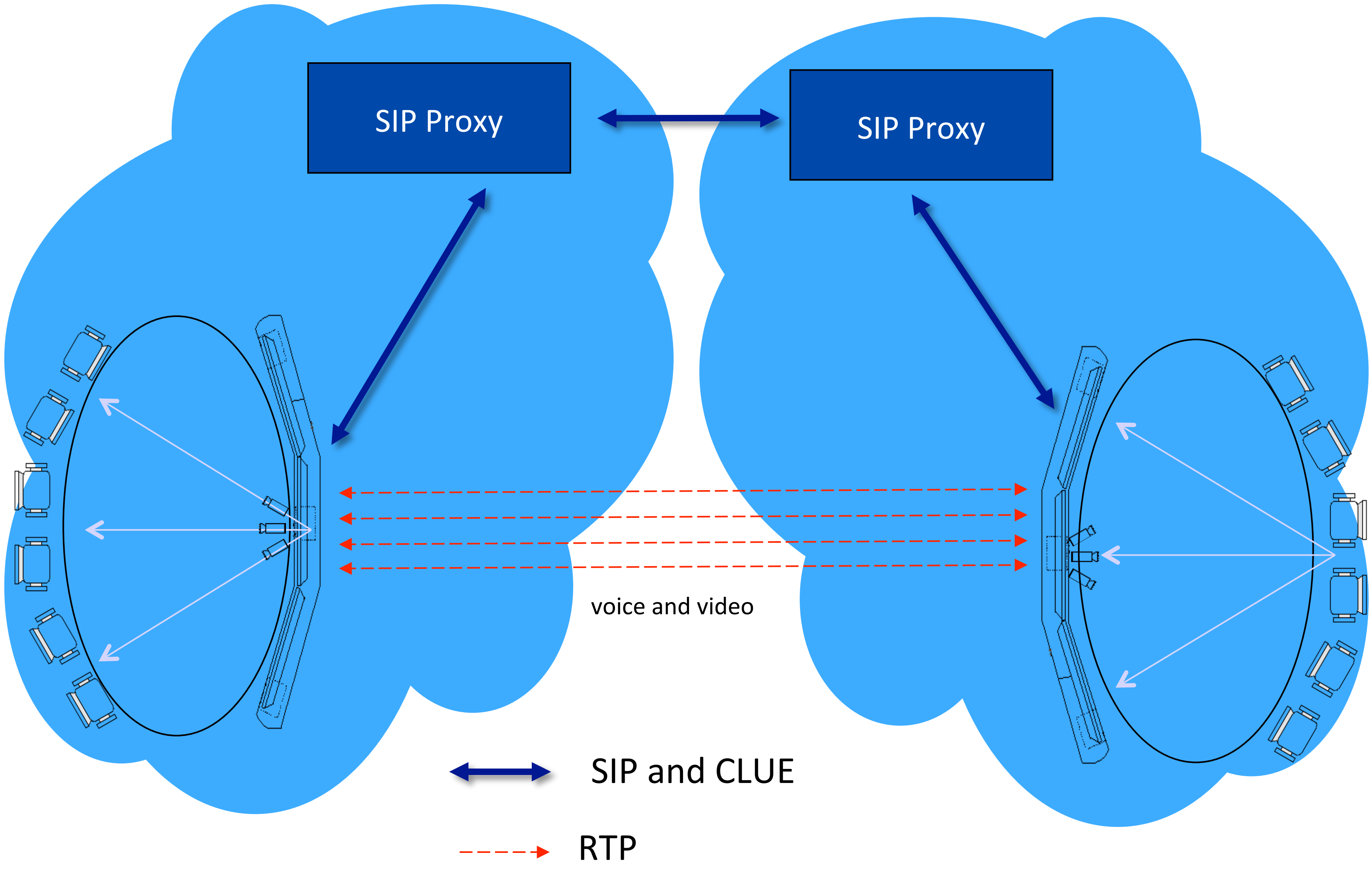
What does RTCWeb do?

- Real-Time Communications in Web Browsers
- Native support in the browser
  - No need for plug-ins
- Browsers download javascript-based real-time applications from web servers using HTTP
- Encrypted RTP is used to transport real-time media between browsers
- SCTP (Stream Control Transmission Protocol) is used for direct browser-to-browser data (e.g. for real-time gaming)
- APIs developed by W3C WebRTC group

Telepresence

- CLUE WG
  - ControLling mUltiple streams for tElepresence
  - Immersive experience
  - Like "being there"
- Conferencing systems with multiple cameras, microphones, and screens
  - Ability to scale images to true size
  - Gaze direction and eye contact
  - Spatial audio

Telepresence



Presence and Messaging

- Presence "state" describes a user's ability and willingness to communicate.
- Examples:
  - What communication mechanisms do I prefer right now?
  - Am I too busy for non-urgent matters?
  - Am I in a quiet environment?
  - Am I engaged in some activity that affects communication?

Location/Privacy

- Let an endpoint learn its geographic location
  - HTTP-Enabled Location Delivery (HELD)
  - DHCP Extensions
- Let an endpoint tell another element/application where it is.
  - Location Conveyance in SIP, HTTP or other protocols
- Provide policy on who can ***see*** that location and what anyone who sees it can do with it.
  - The Privacy part of Geopriv – location comes with rules
- Find available services based on current location
  - Location to Service Translation (LoST)

Calling Party Identity Identity

- Like email, SIP "From" is easily spoofed.
- SIP is a large part of the public telephone network now, and the ability to spoofed caller ID is becoming problematic.
  - Exploits include robocalls, voicemail hacking, bank authentication schemes.
  - Drawing policy attention from, e.g., FCC and ITU
- New work underway in STIR (Secure Telephone Identity Revisited) to tackle this problem specifically for phone numbers, to give providers tools for validation of calling party identity.

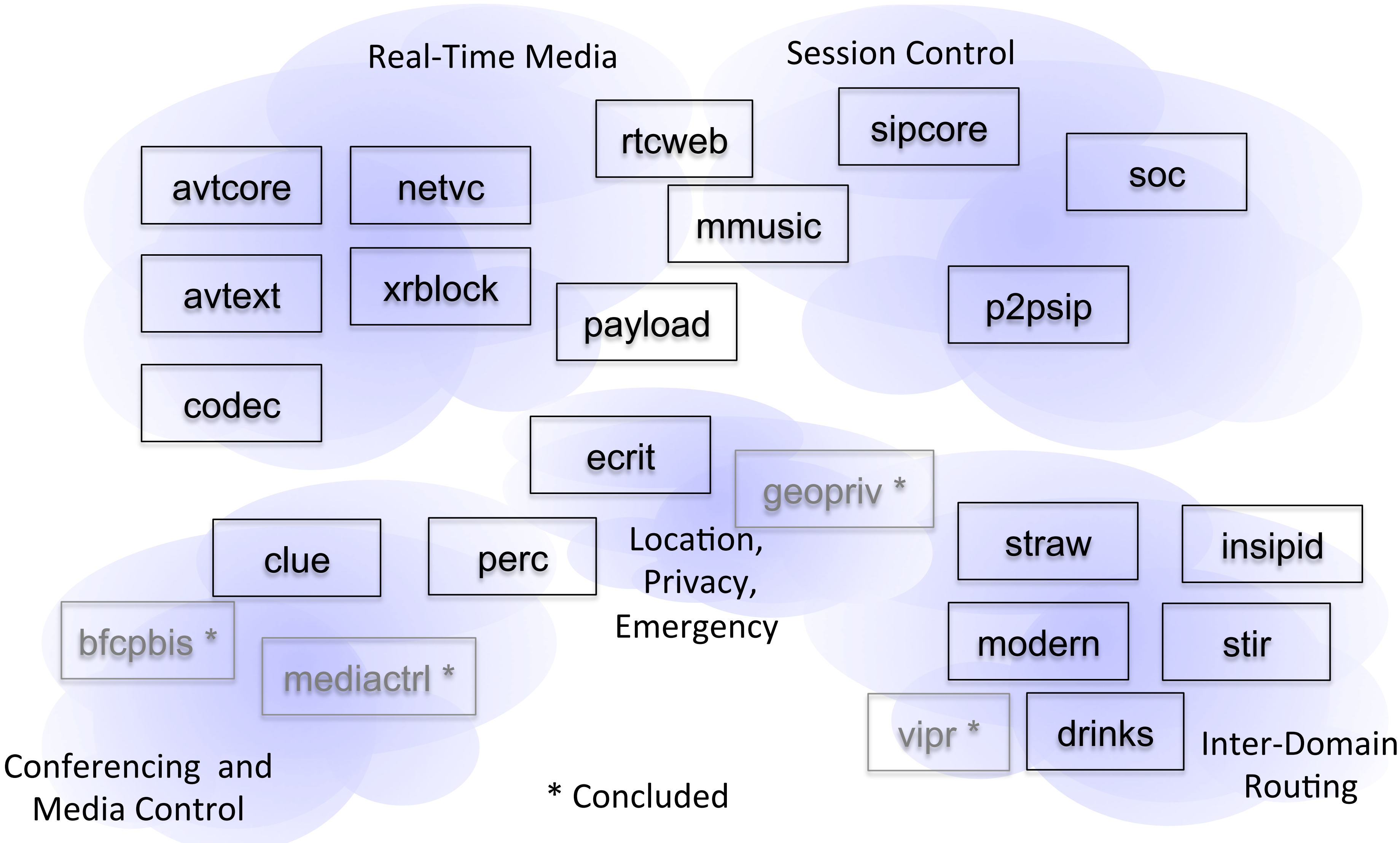
Emergency Services

- Provide the ability to reach the ***right*** emergency responder for the situation
- Provide that responder with the best information for response (location)
- Address legacy and next generation service requirements
  - call-back from the responding service

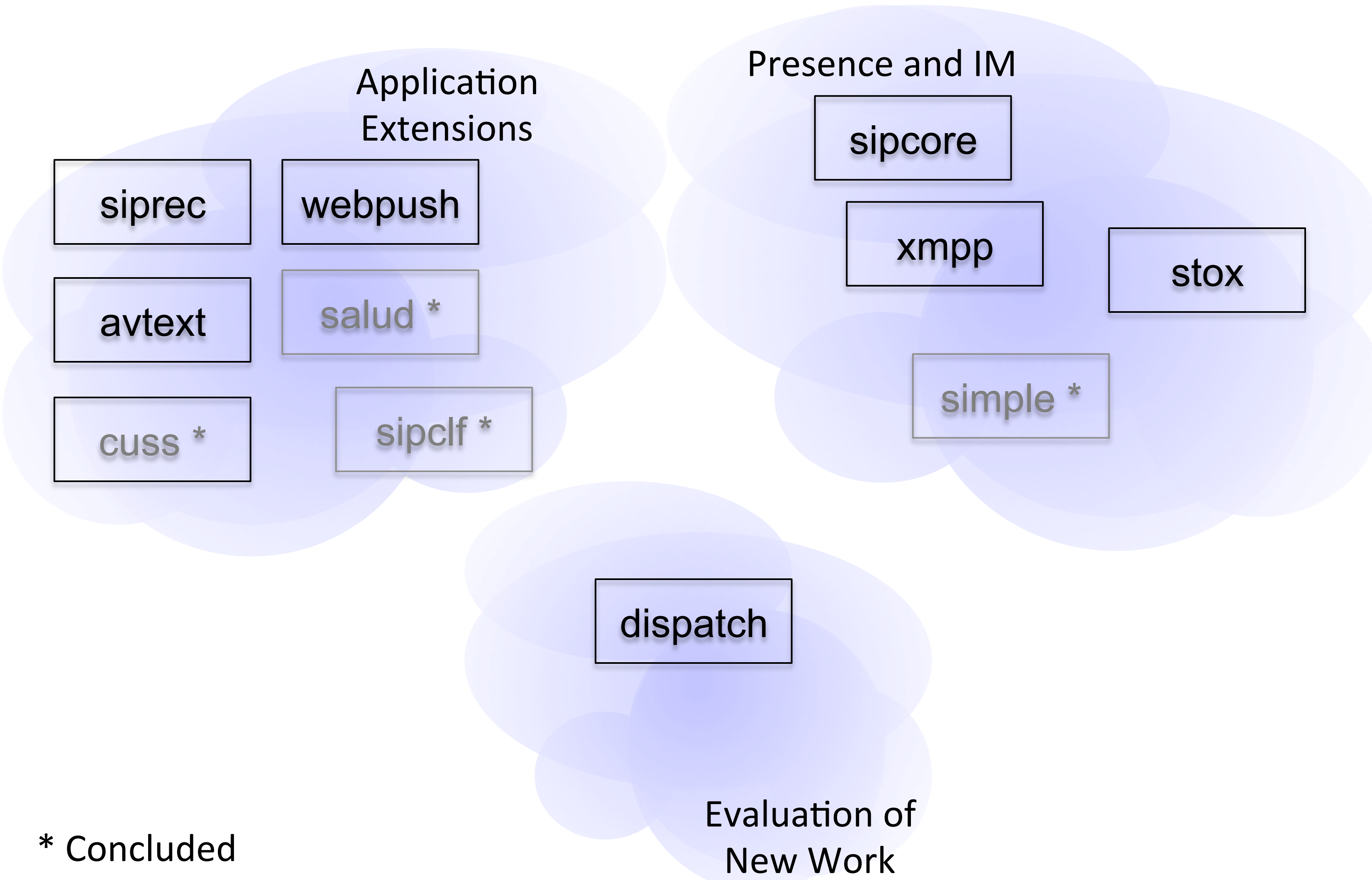
DISPATCH Working Group

- Helps find the right home for new proposed work
  - This is the place to start with a new idea in RAI
  - Dispatches work to an existing working group
  - Helps create a charter for a new group focused on the proposal
  - Makes explicit decisions to not pursue a proposal
- Does not produce protocol documents

Map of Working Groups



Map of Working Groups (cont)



BACKGROUND

WG Overview
Real-Time Media

- avtcore Audio/Video Transport Core Maintenance
- avtext Audio/Video Transport Extensions
- codec Internet Wideband Audio Codec
- netvc Internet Video Codec
- payload Audio/Video Transport Payloads

New

- rtcweb Real-Time Communication in WEB browsers
- xrblock Metric Blocks for use with RTCP's Extended Report Framework

WG Overview
Session Control

- p2psip — Peer-to-Peer Session Initiation Protocol
- mmusic — Multiparty Multimedia Session Control
- sipcore — Session Initiation Protocol Core
- soc — SIP Overload Control
- straw — Sip Traversal Required for Applications to Work
- insipid — INtermediary-safe SIP session ID

WG Overview
Location, Privacy, Emergency Services

- ecrit — Emergency Context Resolution with Internet Technologies
- geopriv — Geographic Location/Privacy *Concluded*

WG Overview
Application Extensions

- cuss Call Control UUI Service for SIP *Concluded*
- salud Sip ALerting for User Devices *Concluded*
- sipclf SIP Common Log Format *Concluded*
- siprec SIP Recording
- webpush Web-Based Push Notifications

WG Overview
Interdomain Routing

- drinks Data for Reachability of Inter/tra-NetworK SIP

  Concluding Soon

- vipr Verification Involving PSTN Reachability

  Concluded

- stir Secure Telephony Revisited

- modern Managing, Ordering, Distributing, Exposing, & Registering Telephone Numbers *New*

WG Overview

Presence and IM

- simple — SIP for Instant Messaging and Presence Leveraging Extensions — *Concluded*
- xmpp — Extensible Messaging and Presence Protocol — *Concluding Soon*
- stox — Sip-TO-Xmpp interoperation — *Concluding Soon*

WG Overview

Conferencing, Telepresence, Media Services

- bfcpbis Binary Floor Control Protocol Bis

  Concluding Soon

- clue ControLling mUltiple streams for tElepresence

- mediactrl Media Server Control *Concluded*

- perc Privacy Enhanced RTP Conferencing *New*

WG Overview
Evaluating New Proposals

- dispatch