Introduction to the Real-Time Applications and Infrastructure Area in the IETF

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What is the area about?

- Tools for letting people interact with each other with minimal delay using the Internet
  - Talking
  - Two- (or more) -way video
  - Gaming
  - Live collaborative music
  - Instant Messaging

Delay Sensitive Interpersonal Communication

What is the area about?

- Building blocks for real-time services
  - Providing (and protecting) location
  - Advertising available real-time services
  - Getting emergency calls to the right responder
  - Allowing applications to react to a person's changing ability or willingness to communicate

What's in the name?

Delay Sensitive
Interactive
Communication

Moving secure voice and video
Providing location

Real-Time Applications and Infrastructure

Internet Telephony
Collaborative Performance
IM and Presence
Emergency Services

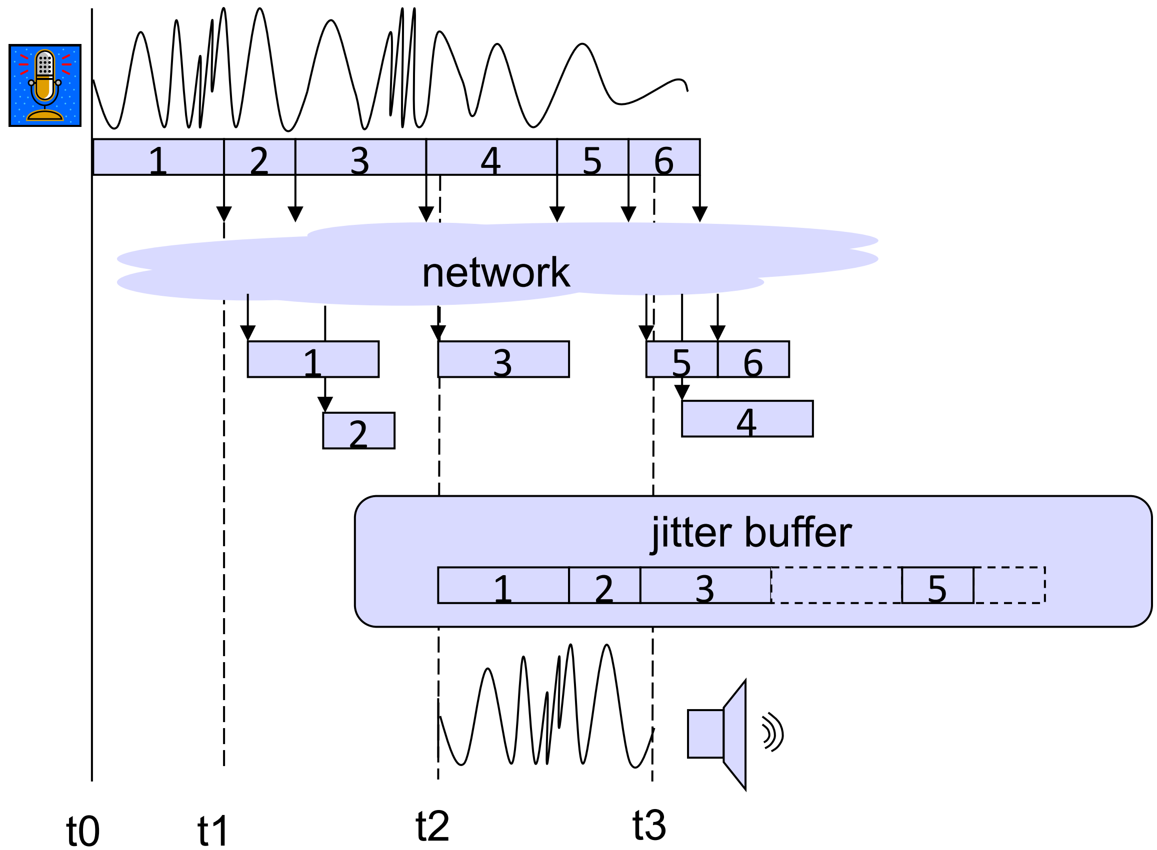
RAI is pronounced the same as "Rye"

In today's overview

- Moving real time media around (RTP)
- Setting up communication sessions (SIP)
- Talking about those sessions (SDP)
- Presence/Messaging (SIMPLE, XMPP)
- Location, Privacy, and Emergency Services

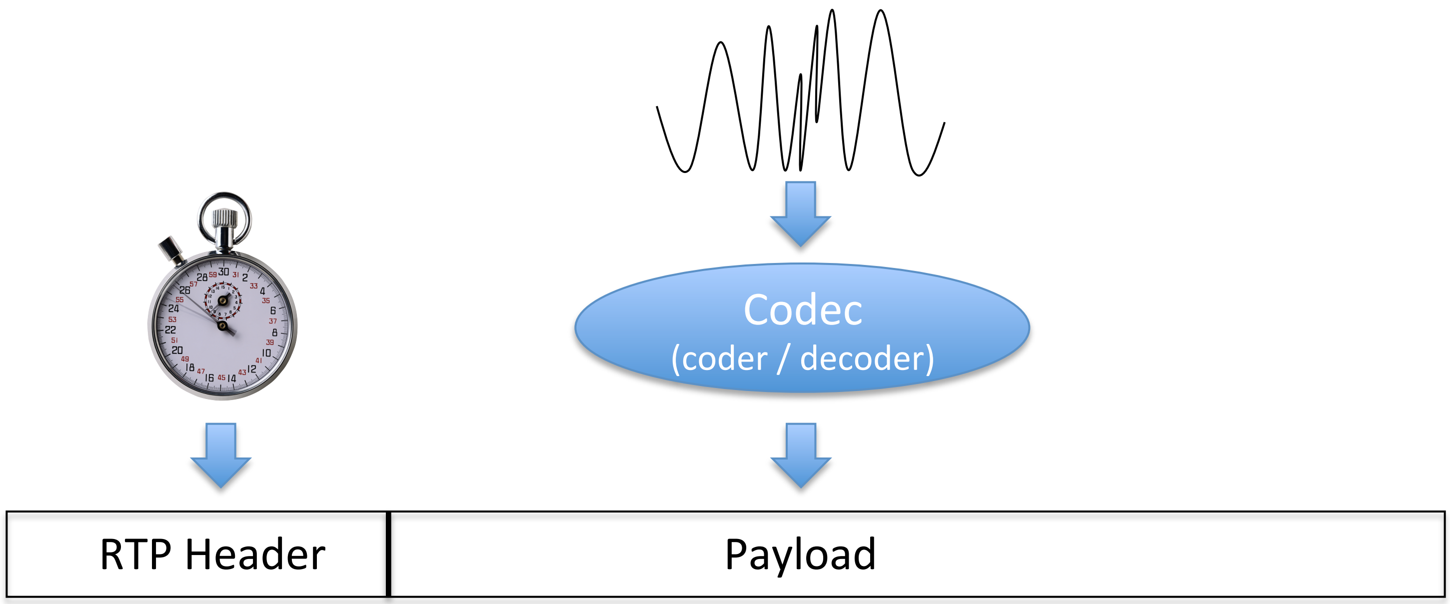
What does RTP do

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



What does RTP carry

- Signals encoded by codecs
- Timed-information directly encoded into the payload (avt-tones, midi)



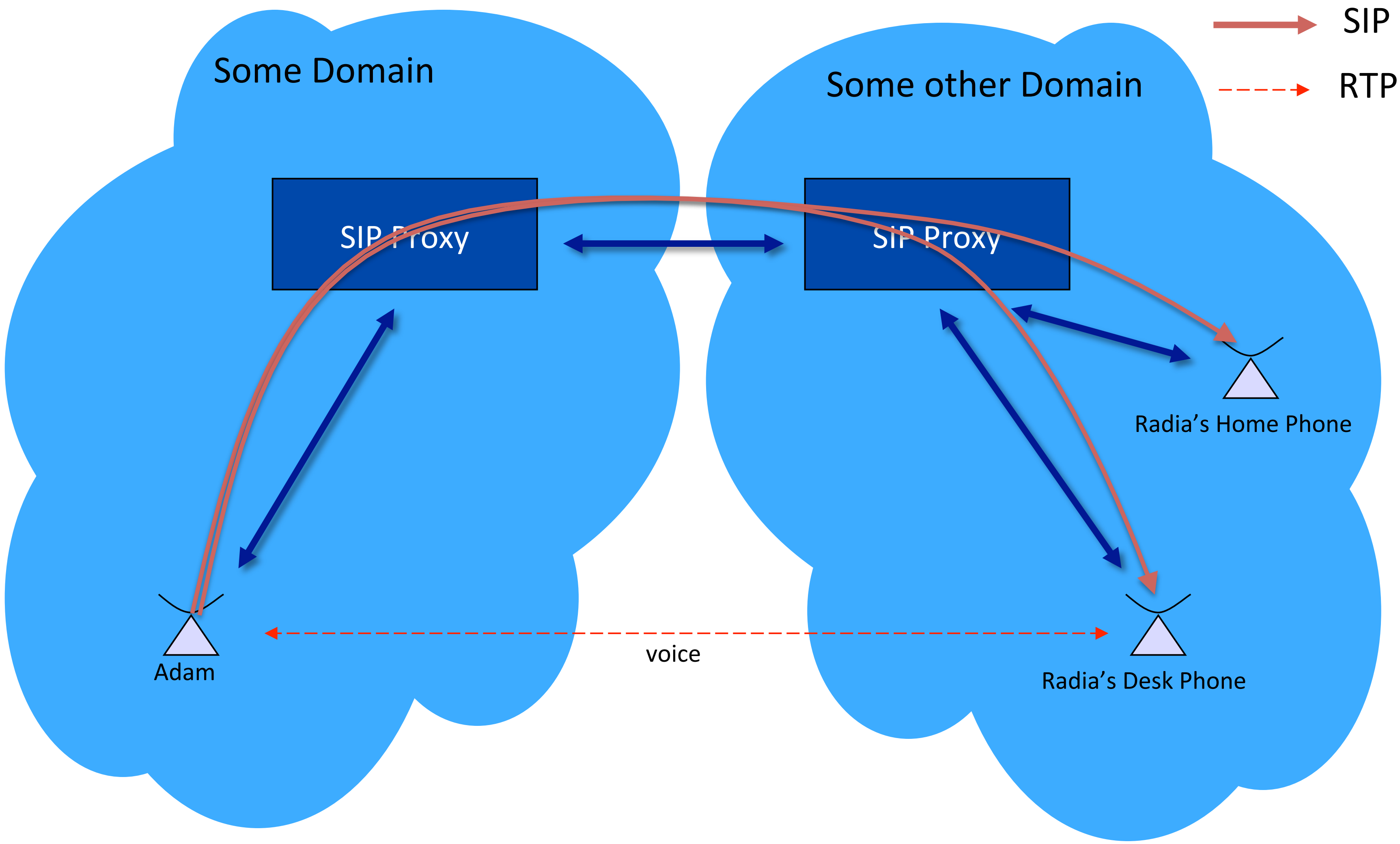
What does SIP do?

- 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.

Finding "the right" Device

- Generally done at the discretion of the called party's SIP servers, using implementation-specific business logic.
- Can include "parallel" alerting (all devices at once), "serial" alerting (one device at a time), or hybrid of the two approaches.
- Some standardized tools defined to help.
  - Callee capabilities/caller preferences mechanism can express things like "this device can do video" when a phone registers, lets caller say "I want to call a video-capable device" when making a call
  - Presence documents can express preferences and capabilities as well.

What does SIP do?

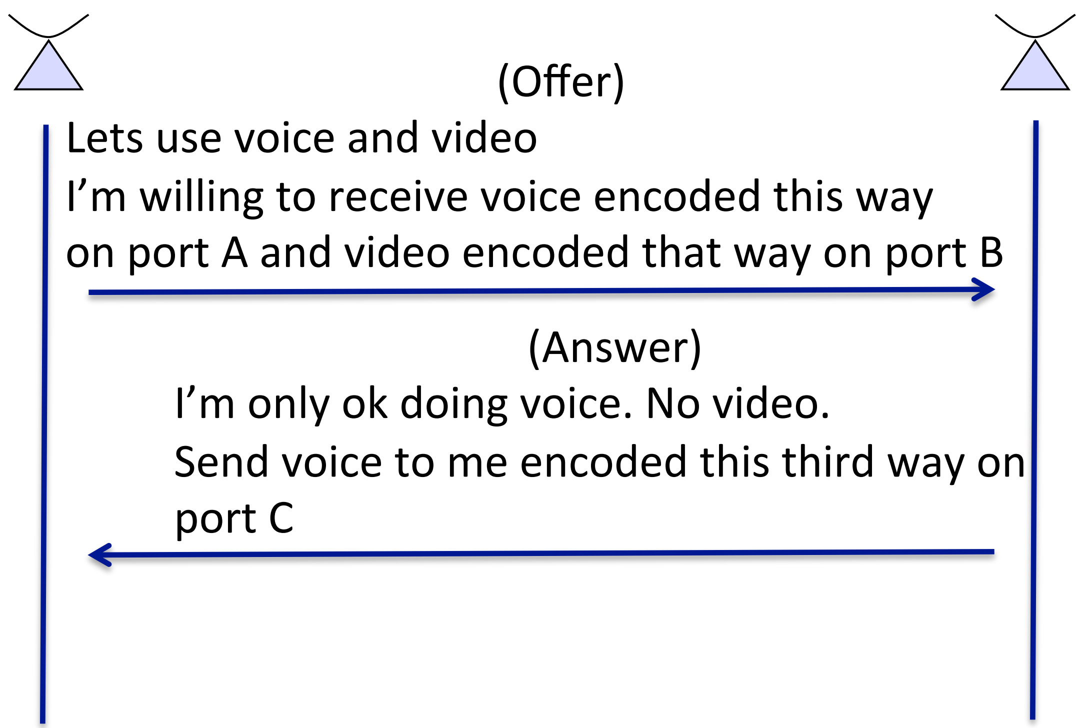


Session Description Protocol (SDP)

- 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

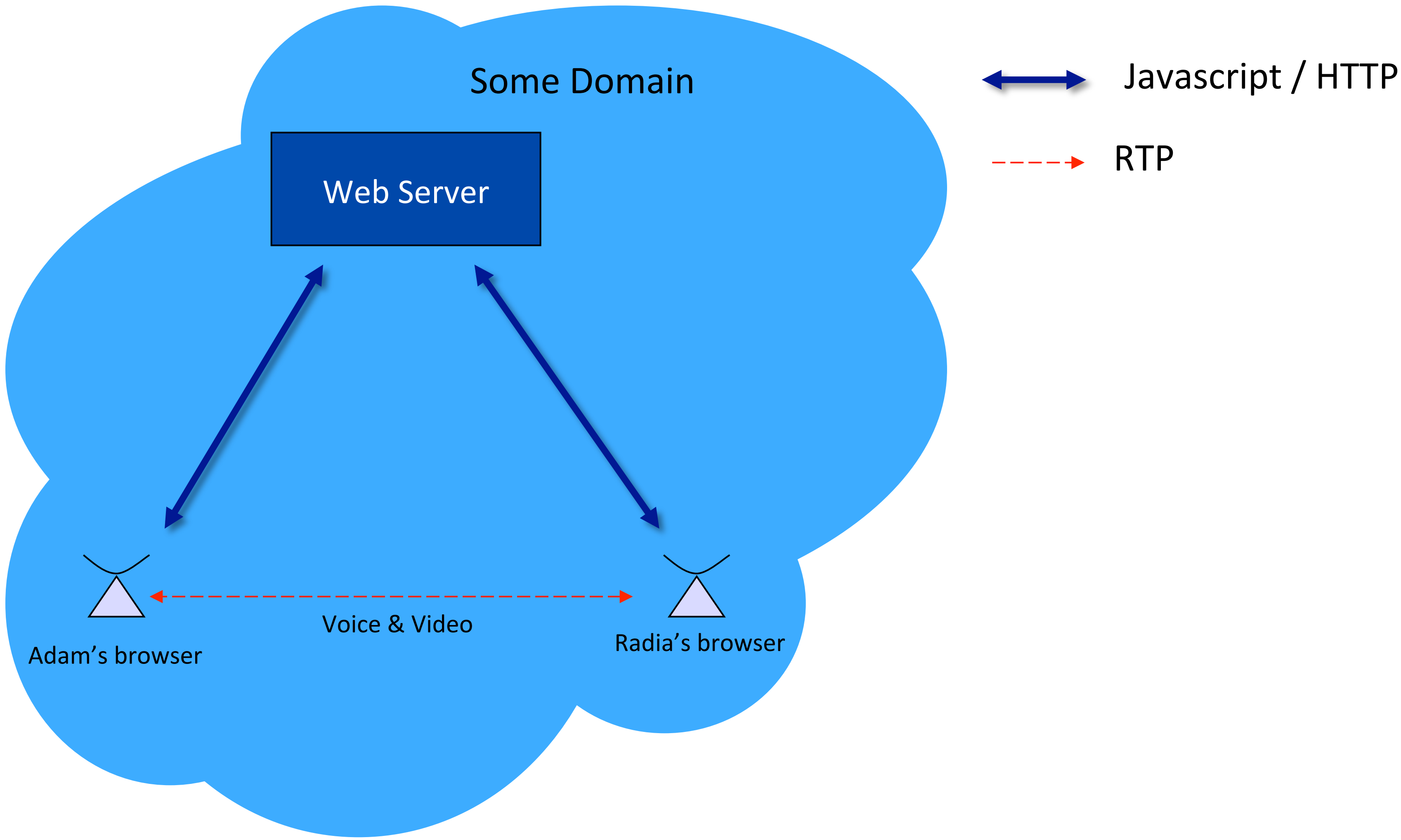
- SIP Devices use SDP to negotiate how to communicate



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

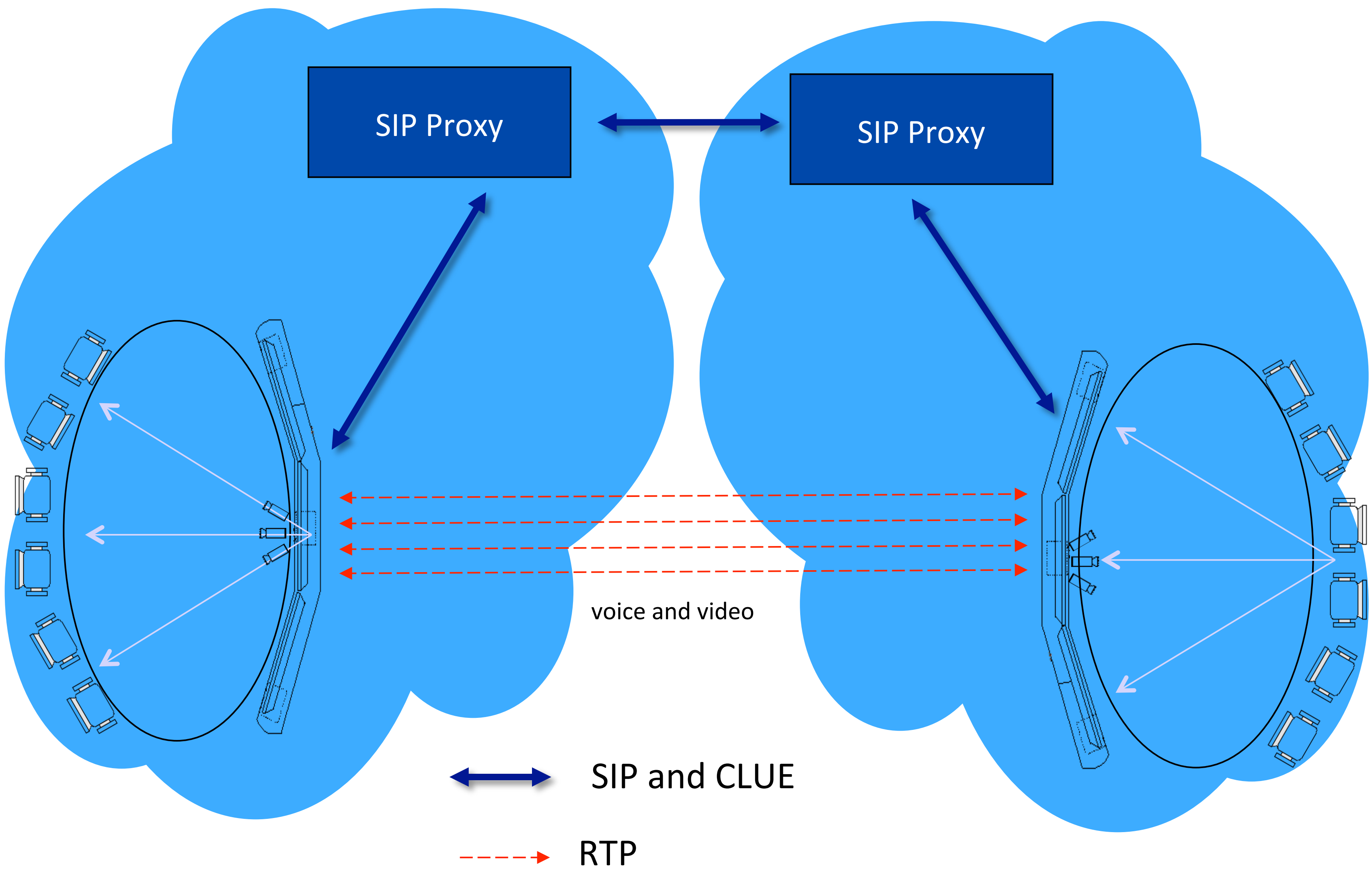
What does RTCWeb do?



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



The pressure RTCWeb and CLUE are putting on the use of SDP and RTP

- Multiplexing
- Mandatory-to-implement audio and video codecs
- Simulcast
- Use of codecs with different clock rates in a media stream
- Congestion control and circuit breakers for real-time media
- Describing relationships among RTP streams and groups

Presence and Messaging

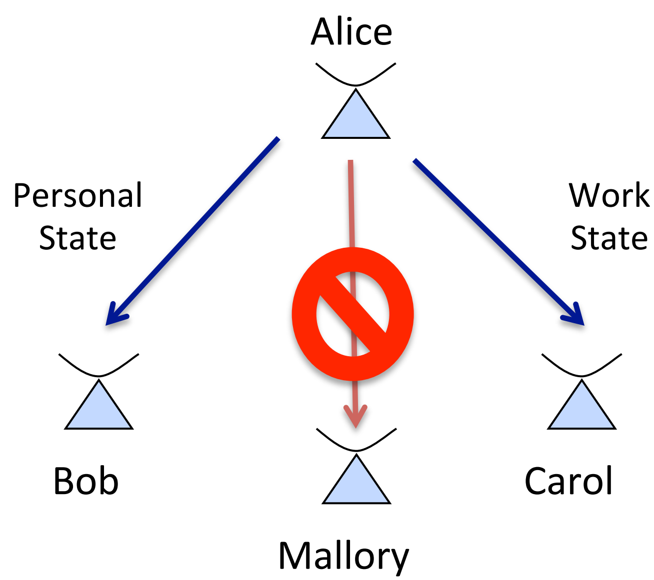
- Presence “state” describes a user’s ability and willingness to communicate.
- Examples:
  - What communication mechanisms do I prefer right now?
  - Am I too busy for non-urgent matters?
  - Am I in a quiet environment?
  - Am I engaged in some activity that affects communication?

Presence State

- Presence State can be a combination of soft and hard state
  - At lunch for the next hour
  - On holiday until I tell you otherwise

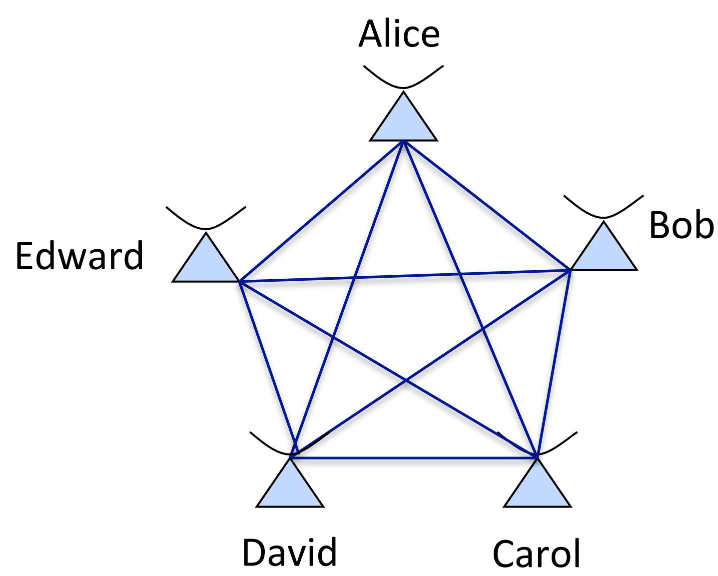
Presence State Distribution

- A presence system distributes state to authorized watchers
  - Different watchers may see different state



Contact Lists

- Distribute presence state to many
- Gather it from many
  - aka buddy lists or rosters
  - Number of relationships scale up quickly.



Messaging

- Several kinds of messaging
  - Page Mode – Short, usually text. Similar to text paging or SMS
  - Session Mode – Chat session with a clear beginning and end
  - Multi User Chat
- Messages can carry arbitrary kinds of content
  - Including transfer of large content; e.g., file transfer

IETF Presence and Messaging Efforts

- **Extensible Messaging and Presence Protocol** (XMPP)
  - Based on XML streams
  - Client-server architecture, with server to server federation
  - Well deployed
- **SIP for Instant Messaging and Presence Leveraging Extensions** (SIMPLE)
  - Primarily SIP based, but includes other protocols (e.g. XCAP, MSRP)
  - Highly flexible architecture (with resulting deployment complexity)
  - Fewer deployments, but starting to grow
- **SIP-to-XMPP Interoperation** (STOX)
  - New working group chartered to publish documents that detail how to interoperate Presence & IM between SIP and XMPP
  - Based on long-standing series of individual documents

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
- Some existing work already in this space:
  - **RFC 3325** adds proxy-controlled ID, but relies on specific architectures.
  - **RFC 4474** allows proxies to sign "From" for their domain, but this doesn't work for phone numbers.
  - **VIPR** establishes identity for repeated SIP calls; but it doesn't hinder robocalling.
- 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

WORKING GROUP OVERVIEWS

WG Overview
Real-Time Media

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

WG Overview
Application Extensions

- cuss Call Control UUI Service for SIP *Concluding Soon*
- salud Sip ALerting for User Devices *Concluding Soon*
- sipclf SIP Common Log Format *Recently Concluded*
- siprec SIP Recording

WG Overview
Interdomain Routing

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

  Concluding Soon

- vipr Verification Involving PSTN Reachability

  Concluding Soon

- stir Secure Telephony Revisited *New*

WG Overview
Presence and IM

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

WG Overview

Conferencing, Telepresence, Media Services

- bfcpbis Binary Floor Control Protocol Bis

Concluding Soon

- clue ControLling mUltiple streams for tElepresence

- mediactrl Media Server Control

WG Overview
Evaluating New Proposals

dispatch