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?

Real-Time Applications and Infrastructure

Delay Sensitive Interactive Communication

Moving secure voice and video Providing location

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?

Some Domain

Adam

SIP Proxy

Some other Domain

Radia’s Desk Phone

Radia’s Home Phone

SIP

RTP

voice
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
• SIP Devices use SDP to negotiate how to communicate

(Offer)

Lets use voice and video
I’m willing to receive voice encoded this way on port A and video encoded that way on port B

(Answer)

I’m only ok doing voice. No video.
Send voice to me encoded this third way on port C
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?

Some Domain

Web Server

Voice & Video

Adam’s browser

Radia’s browser

Javascript / HTTP

RTP
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

SIP Proxy

SIP Proxy

voice and video

SIP and CLUE

RTP
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

• vipr    Verification Involving PSTN Reachability

• stir    Secure Telephony Revisited

Concluding Soon

Concluding Soon

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
- clue  Controlling multiple streams for Telepresence
- mediactrl  Media Server Control
WG Overview
Evaluating New Proposals

dispatch