RTCWEB interim 2011-09-08
Remote recording use case / requirements

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Background

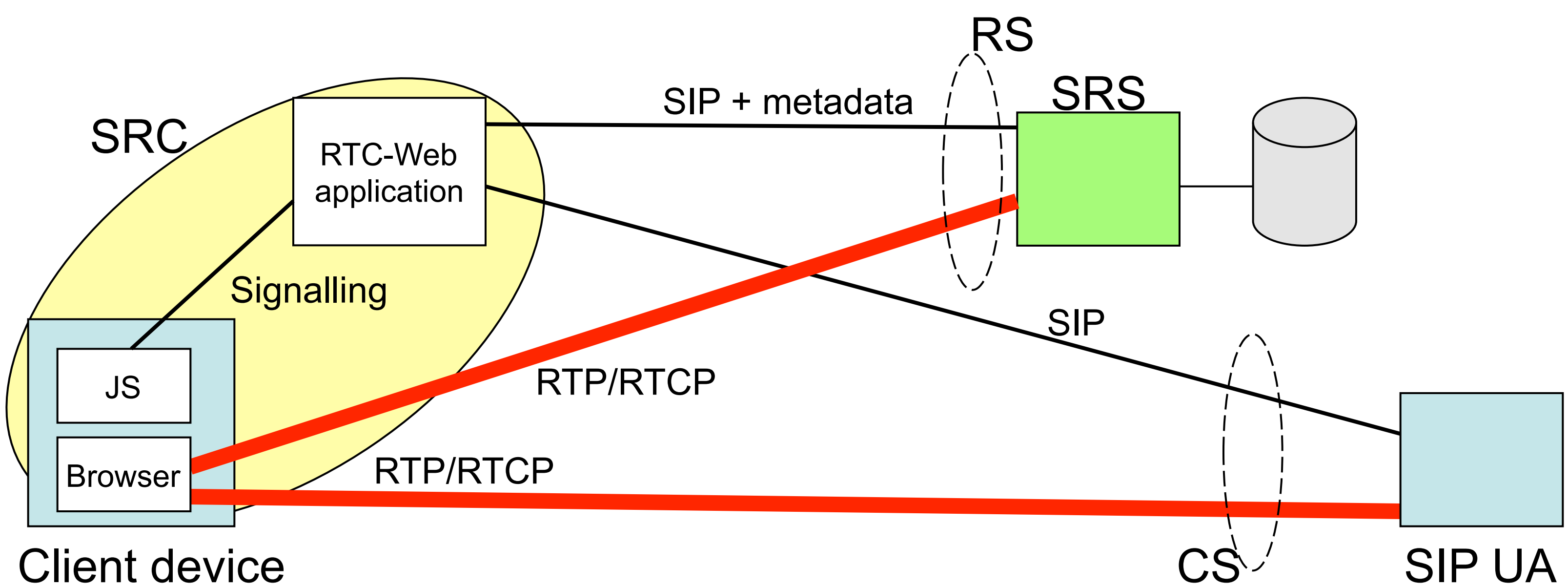
- The SIPREC WG is specifying the use of SIP for delivering real-time media and metadata from a communication session to a recording device (remote recording)
  - From Session Recording Client (SRC) to Session Recording Server (SRS)
- Many business applications, e.g., in contact centres, where recording is needed for legal, non-repudiation, quality control and other reasons.
- Active recording – with knowledge / implicit consent of participants
- More background to SIPREC in back-up slides
- How is session recording achieved in an environment using RTC-Web clients?
- Out of scope for this presentation:
  - Local recording (on client device)
  - Voicemail / videomail

SIPREC and RTC-Web (1)

- There will be some environments that deploy RTC-Web clients and require communication sessions to be recorded centrally (SIPREC)
- Can locate the SRC at a middlebox (e.g., a SIP B2BUA), as will be done in many SIP deployments
  - Requires CSs to be routed through the middlebox concerned (if not already)
  - Requires a standardized signalling protocol between RTC-Web application and the middlebox (e.g., SIP)
  - In conflict with peer-to-peer principle of RTC-Web real-time media
  - Will satisfy some deployments

SIPREC and RTC-Web (2)

- Or can have a decomposed SRC implemented partly in the application and partly in the browser
  - SIP and metadata aspects in the application
  - Media aspects in the browser



Browser support for remote recording

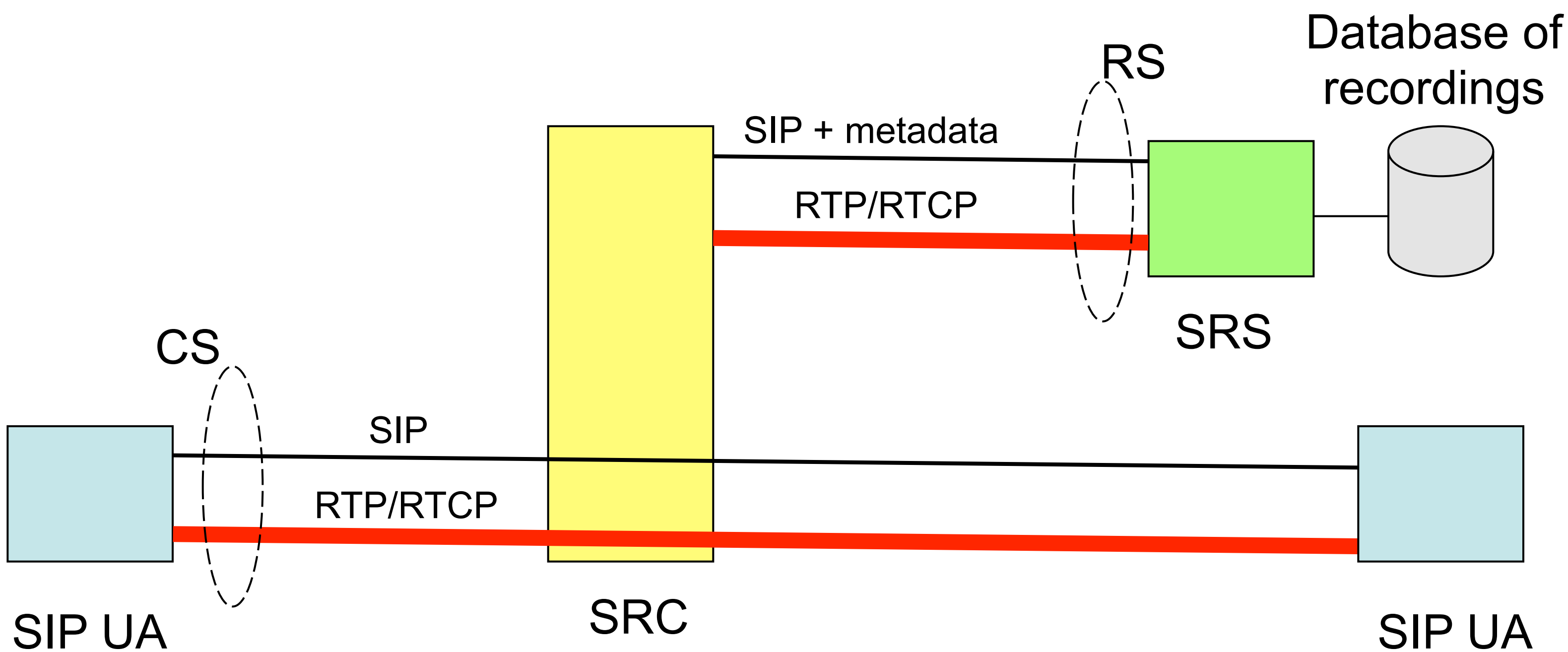
- Main requirement:
  - Copy streams sent/received on one peer connection and send them on another peer connection to the SRS
- Application responsibility
  - SIP support for SIPREC (assuming SIP remains outside the browser)
  - Gathering and sending of metadata to SRS
  - Control of tones / announcements / notifications
  - Control of the two peer connections
- Issues with this approach (to be considered during work)
  - Bandwidth considerations on the local link, particularly uplink (this is a concern for any UA implementation of a SIPREC SRC)
  - Security issues in using device as a media relay - ensuring appropriate permissions given

Questions for the RTCWEB WG

- Is a middlebox approach to session recording sufficient for RTC-Web environments, or do we need support at the endpoint (i.e., in the browser)?
- Is this support needed at day 1?
- If not needed at day 1, should we ensure the architecture does not prevent support in the future (copying between peer connections without architectural re-design)?
- Should we add a use case and corresponding requirements?

Back-up

Overview of SIPREC (1)



- CS is the Communication Session between two or more participants
- RS is the recording session between Session Recording Client (SRC) and Session Recording Server (SRS) (recording device)

Overview of SIPREC (2)

- Media delivered to SRS in real-time using RTP, allowing applications such as real-time analytics
- Only concerned with real-time media
- The SRC must sit somewhere on the path of the CS with access to signalling and media
  - at a B2BUA such as an SBC
  - at a conference focus
  - at an endpoint (SIP UA)
- Present work assumes SRC is aware of policy concerning which calls / which media to record
- SRC responsible for notifying participants in accordance with policy (e.g., by applying tones or announcements, explicitly signalling to recording-aware UAs)
- Retrieval, playback, retention etc. of recordings out of scope

Requirements for browser support for remote recording

- Copy streams sent/received on one peer connection and send them on another peer connection to the SRS
  - Possible need to mix streams before sending (e.g., mix sent and received audio streams)
  - Possible need to use a different codec or other parameters for sending to the SRS
  - Possible need to use a different encryption / integrity context for sending to the SRS
- Possible need to insert tones or announcements into the original media path being recorded
- Possible need to support SDP enhancements for indicating media being recorded or preferences for media to be recorded

Proposed use case / requirements

- As sent out 2011-08-25, augmented by a proposal from Jim McEachern 2011-08-22:

4.2.yy Remote Session Recording

In this use case, the web application user wishes to record a real-time communication at a remote third party (e.g., a recording device), such that transmitted and received audio, video or other real-time media are transmitted in real-time to the third party. The third party can perform recording, archiving, retrieval, playback, etc., but can also perform real-time analytics on the media. A typical deployment might be in a contact centre. For a given medium, the two directions of transmission can be transmitted together (mixed) or separately. The web application also sends metadata that gives context to the stored media. If required, the web application will direct the browser to insert an appropriate indication (e.g. an intermediate beep) into the media stream to show that the communication is being recorded.

New requirements:

Fyy1: The browser MUST be able to send in real-time to a third party media that are being transmitted to and received from remote participants.

Ayy1: The web application MUST be able to ask the browser to transmit in real-time to a third party media that are being transmitted to and received from remote participants and, in the case of audio at least, ask for the two directions of transmission to be transmitted to the remote recording device mixed or separately.

Ayy2: The web application MUST be able to ask the browser to insert an appropriate indication into the media stream to indicate that the communication is being recorded.