Media

draft-ietf-stox-media

Interim Meeting
17 December 2013

Overview

- ~~Still in very raw state, needs love~~
- ~~Scope was unclear~~
- ~~Scope: basic audio / video call interoperability~~ (done)
  - ~~3264 compatible, extensions later~~ (done)

The Plan

- ~~Reorganize draft structure~~ (done)
- ~~Define mappings for the Call Model and for SDP, separated~~ (done)
- Advanced signaling use cases are out of scope
    - Call transfer, early media, third-party call control, etc.

Open Issues (1)

- ~~How many call-flow scenarios need explaining?~~ (done - just the basics)
- ~~ICE: if 3264 compat is used, ignore transport-info?~~ (done)
- ~~a=fmtp syntax translation~~ (done)
- ~~Mapping of hold: session-info vs content senders~~ (done)
- ~~Add text about trickle ICE~~ (done)

Open Issues (2)

- ~~Need some text about forking~~ (done)
- ~~Need some text for loops / spirals involving endpoints across SIP - XMPP~~ (done)
- ~~Need text stating that early media is out of scope~~ (done)
- Need text stating that call transfer is out of scope

Open Issues (3)

- What to do about "double hold" (each side puts the other on hold)? (a=inactive)
- How to translate SIP dialog IDs into Jingle session IDs and vice-versa? (this is needed to prevent spirals and loops)
- Define support for late offer/answer? (our opinion is "no, thank you")

Open Issues (4)

- Need to add more examples (e.g., for format-specific parameters)
- Mention RTP applications other than voice and video (T.140 text and T.38 fax)
- Add text about intended audience (see AppsDir review of draft-ietf-stox-presence)

Path Forward

- (discussion about how to proceed)