The state of Federated RTC and WebRTC in Debian, 2015

Daniel Pocock
http://danielpocock.com

Before we start..

- Debian Developers:
  - Please create an RTC password for yourself, see https://rtc.debian.org
  - It takes time for new passwords to be provisioned in the SIP server, maybe 15 minutes

What is Federated RTC?

- RTC = Real Time Communications
- Federated: calling from domain to domain
- WebRTC: using browser as client

WebRTC

- Not truly peer-to-peer, minimal server required
- Works well browser to browser
  - All browsers have common codecs
  - All browsers have NAT traversal
- Doesn't work browser-to-softphone without extra help from servers or newer softphones
- Doesn't mandate use of SIP or XMPP
- Some limitations (e.g. moving off page, battery)

How do we federate?

- Using DNS to find NAPTR and SRV records
- Using mutual TLS authentication to prevent spoofing and reduce spam calls
  - Spoof calls more dangerous than spoof emails
- Get started quickly with:
  - Debian packages
  - http://RTCQuickStart.org

From two domains to the world?

- We've proven it works for two domains
- How to go global?
- Can we take on Skype?
- What are the other threats to free/open RTC?
  - Proprietary WebRTC, expect to see services like Facebook, SalesForce and LinkedIn trying to displace Skype

Will organizations choose SIP?

- Motivated by features (e.g. email killing fax because it is faster and more convenient)
- Many FLOSS proponents focus on privacy but not every business is prioritizing the privacy issue, even if it relevant to them.
- The choice between SIP and XMPP has muddied the waters (most deployments must have both these days)
- Therefore:
  - must identify and create compelling features
  - Must ensure decision makers are made aware of the features that they are most likely to value
-

How can FLOSS communities stake their claim on WebRTC?

- Making WebRTC modules for existing frameworks
  - e.g. DruCall for Drupal sites
- Using open standards for the backend
  - e.g. SIP and XMPP
- Getting open source softphones to interact with WebRTC users

Demos

- JSCommunicator
  - http://jscommunicator.org
  - https://freephonebox.net
  - https://rtc.debian.org
- Jitsi Meet
  - http://meet.jit.si

Next steps

- Set it up yourself http://rtcquickstart.org
- Discuss on mailing lists:
  - https://lists.fsfe.org/mailman/listinfo/free-rtc
- Debian Developers
  - try the https://rtc.debian.org portal
  - Invite people to call you from https://freephonebox.net?dial=you@debian.org