Technical overview of SIP-XMPP co-existence

Simo Veikkolainen

IETF#76 Hiroshima, Japan

13 Nov 2009

End-goals

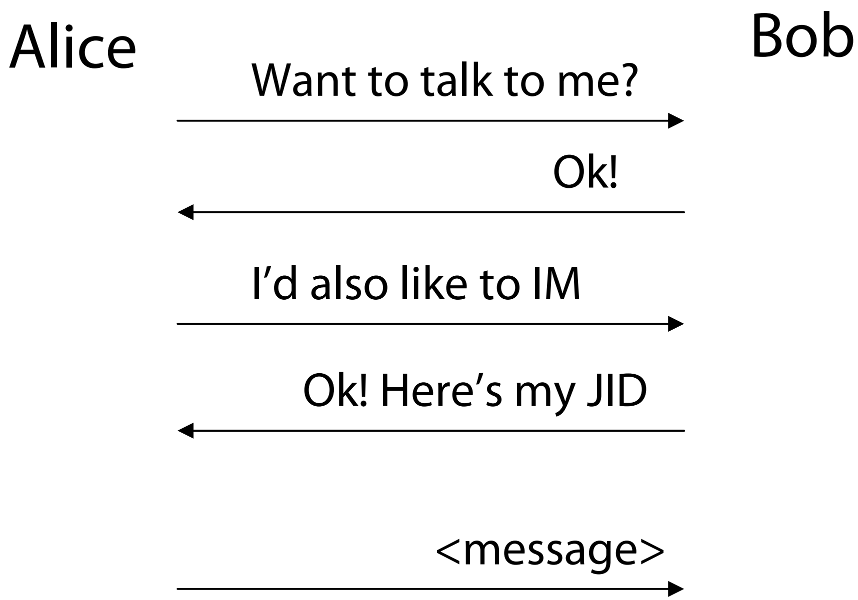
- Provide seamless user experience for voice, IM and presence
- As if using just one protocol
- But in reality using two protocols with independent service infrastructures and addressing
- Only endpoints affected – rapid deployment
- First attempt described in http://tools.ietf.org/id/draft-veikkolainen-sip-voip-xmpp-im-01.txt

Challenges

- Since we are using two different and independent protocols, some glue is needed
  - Discovering address of the endpoint(s) in the other protocol domain
  - Combining the sessions in the endpoints using some correlation mechanism
- Overlapping features (XMPP also does voice, and SIP does IM/Presence)
- Need to make sure we don't break anything that works today

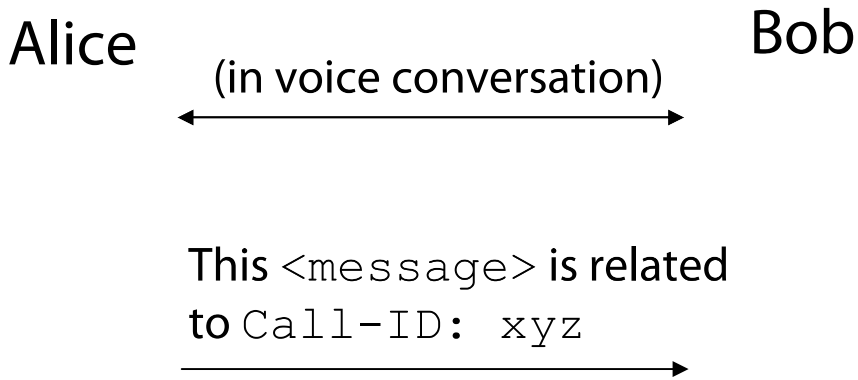
Addressing

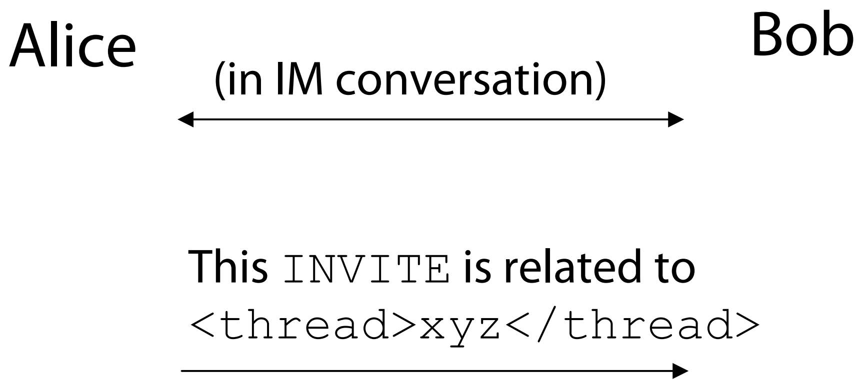
- Need to find out the address in the other domain
    - First a SIP session, add IM
    - First IM, add SIP session
- Addresses may also be configured permanently
    - In a phonebook or directory
- Or you could use XEP-0152
    - Via presence
    - Via Personal Eventing Protocol (PEP)
- Often sip:yukiyo@ex.jp is the same user as xmpp:yukiyo@ex.jp but cannot be relied upon



Correlation

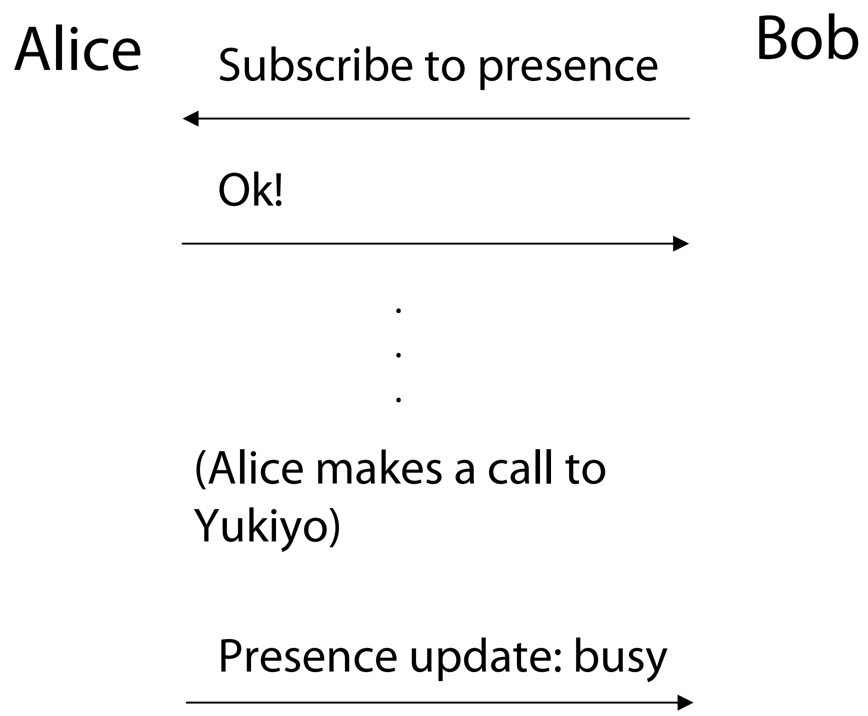
- Need to be able to indicate that
  - A new IM session in related to an ongoing SIP session
  - A new SIP session is related to an ongoing IM session
- Opaque identifiers exist
  - SIP has `Call-ID`
  - XMPP has `<thread>`
  - Use these





Presence

- Relevant information that can be learned through presence for our purposes include
  - Address information
  - Availability information (busy, unavailable, etc.)
- Need more details on this



List of technical issues discussed [1]

- Relationship to SIP-XMPP interworking (draft-saintandre-sip-xmpp-*)
- Video as well as voice? Yes.
- Relationship to Disaggregated media (draft-loreto-dispatch-disaggregated-media)
- Additional features
  - Multiparty
  - SIP Supplementary Services
- Real-time text. Out of scope?
- Call-ID usage

[1] http://www.ietf.org/mail-archive/web/dispatch/current/msg00845.html

Proposal for way forward

- Start with simple use cases
- Extend later
- Process and charter - next

Proposed charter

http://www.ietf.org/mail-archive/web/dispatch/current/msg00560.html

Currently most standards-based Voice over IP (VoIP) deployments use the Session Initiation Protocol (SIP). In addition to providing basic voice service SIP has an extensive support for more advanced telephony features including interworking with the traditional Public Switched Telephone Network (PSTN). SIP is also gaining popularity in the field of video communication. At the same time, the Extensible Messaging and Presence Protocol (XMPP) is enjoying widespread usage in instant messaging and presence services.

Both SIP and XMPP offer extensions for voice as well as IM and presence (XMPP voice via the Jingle extension, and SIP IM/presence via SIMPLE protocols). However, widespread deployment of these extensions has not so far taken place. In order to speed up deployment of integrated VoIP and IM systems, SIP based voice and XMPP based IM/Presence could be combined in endpoints to offer a voice, IM and presence service without any changes to existing SIP and XMPP service infrastrucure.

The objective of this Working Group is to develop solutions for combining SIP based voice with XMPP based IM and Presence such that a unified user experience can be offered to end user. Specifically, solutions are needed on

- addressing; end users should be able to initiate sessions to a user identity in either SIP or XMPP domain. The corresponding user identity in the other protocol domain needs to be learned automatically.
- session correlation; endpoints need to be able to correlate voice sessions with IM/Presence such that they can be presented to the end user in a seamless fashion
- presence; it should be possible to change the XMPP presence status based on the user's activity in the SIP domain.

The goal is to rely on existing service infrastructre, and new requirements should be imposed only to the endpoint.

Protocol interworking, that is, translation from one protocol to another, is out of scope of this WG.

Milestones

Feb 2010 Problem statement and use cases submitted to IESG as Informational

Dec 2010 Specification on combining SIP based voice and XMPP based IM/Presence in a seamless manner submitted to IESG as Proposed Standard.