Farsight 2
Videoconferencing made easy

Olivier Crête

Collabora

Origins of Farsight

- aMSN
- Free IM had no VoIP
- Each proprietary IM protocol had its own thing
- Philippe's end of studies project
- Hacked into aMSN, gaim

Original goals of Farsight

- Enable Free Software IM to do audio/video like other platforms
- Abstract the streaming from different IM protocols
- Hide the complexities of media streaming (of GStreamer)

State of Farsight

- Current used in Telepathy Stream Engine
- On Nokia Internet Tablets
- Very stable
- Thin core, complex plugins
- Complex RTP plugin
- Unmaintained plugins for MSN, Yahoo

RTP plugin

- 1 to 1 audio & video calls
- Codec detection, negotiation
- Transmitters plugins
  - Unicast UDP
  - ICE (libjingle)
- DTMF
- Confort Noise (on Nokia Tablets)
- RTCP

Limitations

- Only one to one calls
- No lip-sync
- Video support broke abstraction
- Hard to use with non-trivial GStreamer pipelines
- No sRTP
- Hacks for Nokia Tablets (DSP, CN)

Farsight 2: Goals

- High level objects
- Interface, helper libraries
- RTP is reference, most standard, most capable
- Also, MSN, Yahoo, etc
- One GStreamer element per protocol
- Elegance
- Automated test coverage
- Good documentation

New RTP plugin

- Keep good things from older versions
  - Codec detection
  - Codec negotiation
  - gst elements: DTMF, CN, RTP payloaders, etc
- Use GStreamer rtpmanager
  - Multi-party
  - Lip-sync
  - Complete RTP feature set
    - Including full RTCP, SSRC collision detection, etc

Transmitters

- Multi Unicast UDP (with STUN)
- Multicast UDP
- Interactive Connection Establishment (ICE)
- Pidgeons, etc

High level objects

- Codec
- Candidate
- Participant
  - One person with synchronized streams
- Session
- Stream
- Conference

Collabora

Session

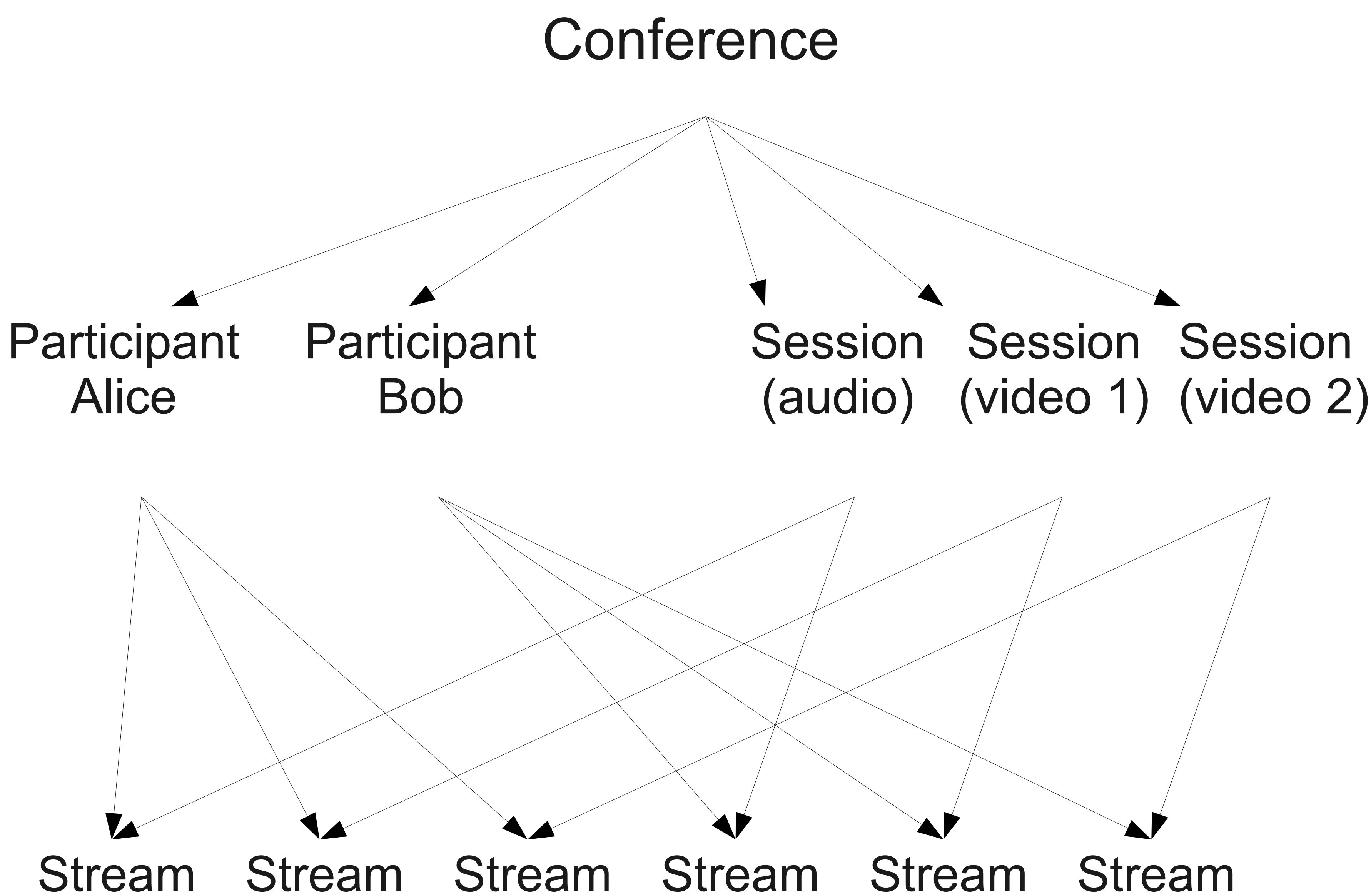
- One type of media (audio, video, etc)
- One local media source
  - One microphone
  - One camera
  - File
  - etc
- Multiple stream from other participants
- RTP session

Stream

- One participant in one session
- Use for communication with participant
  - Codecs
  - Candidates
- Remote media comes out of here

Conference

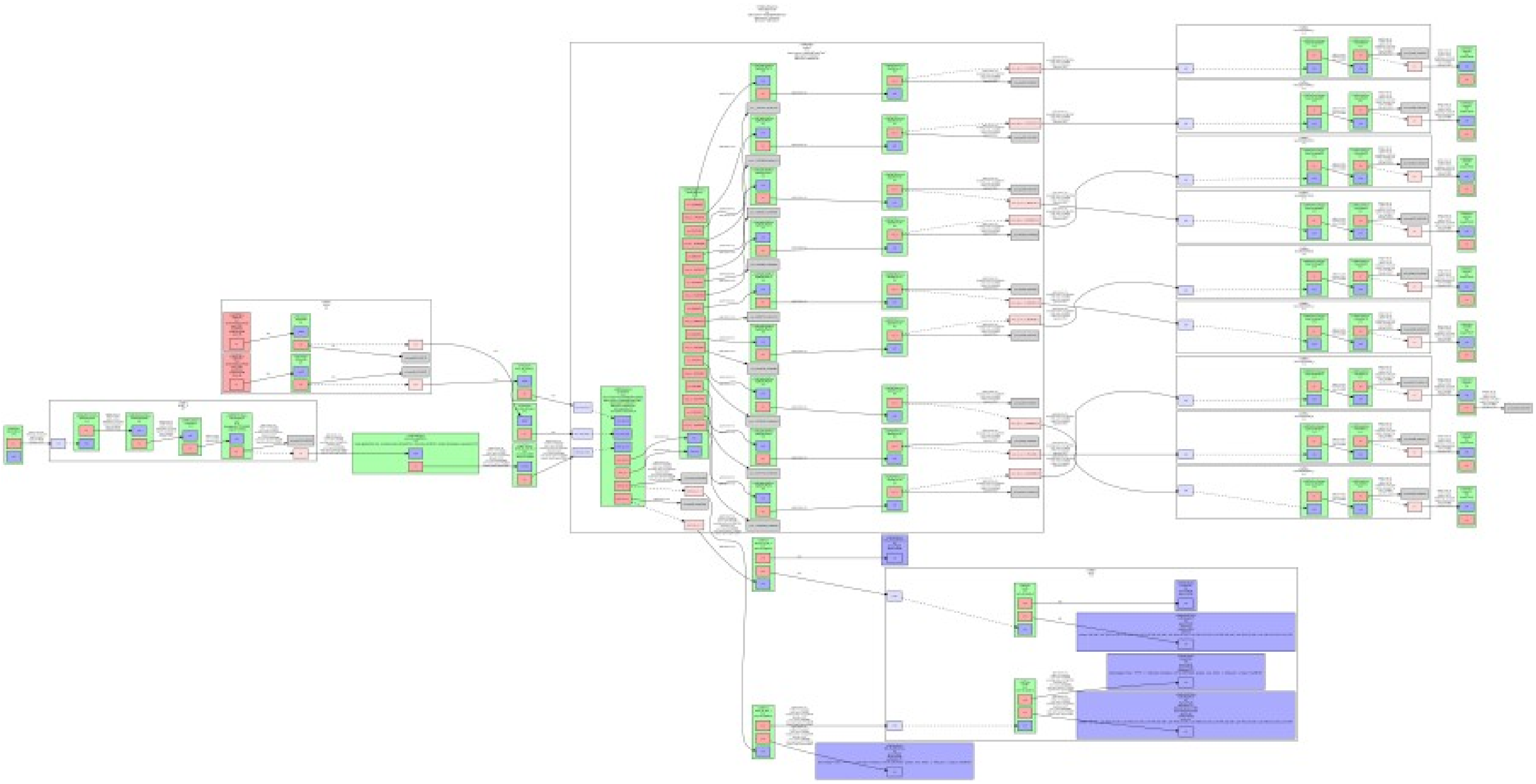
- The GStreamer element
- Multiple synchronized sessions
- Contains everything else


Conference
Participant Alice
Participant Bob
Session (audio)
Session (video 1)
Session (video 2)
Stream
Stream
Stream
Stream
Stream
Stream

Current status

- Base RTP implementation
  - Most of Farsight 1 features, except DTMF & CN
  - Multi-party
  - Lip-sync
  - Python bindings
  - Some automated tests
  - Unicast, Multicast transmitters

10 way conference



Example

```
import farsight, gst, gobject

loop = gobject.MainLoop()
pipeline = gst.Pipeline()

conference = gst.element_factory_make ("fsrtpconference")
conference.set_property ("sdes-cname", "tester@2.3.4.5")
pipeline.add (conference)

session = conference.new_session (farsight.MEDIA_TYPE_AUDIO)
participant = conference.new_participant ("bob@1.2.3.4")
stream = session.new_stream (participant, farsight.DIRECTION_BOTH, "multicast")
```

```
stream.set_remote_codecs(session.get_property("local-codecs"))

candidate = farsight.Candidate()
candidate.ip = "224.0.0.110"
candidate.port = 3442
candidate.component_id = farsight.COMPONENT_RTP
candidate.proto = farsight.NETWORK_PROTOCOL_UDP
candidate.type = farsight.CANDIDATE_TYPE_MULTICAST
stream.add_remote_candidate (candidate)

candidate.port = 3443
candidate.component_id = farsight.COMPONENT_RTCP
stream.add_remote_candidate (candidate)
```

```
audiosource = gst.factory_element_make ("audiotestsrc")
pipeline.add (audiosource)
audiosource.get_pad ("src").link(session.get_property ("sink-pad"))

def _src_pad_added (stream, pad, codec, pipeline):
    audiosink = gst.element_factory_make ("alsasink")
    pipeline.add (audiosink)
    audiosink.set_state (gst.STATE_PLAYING)
    pad.link (audiosink.get_pad ("sink"))

stream.connect ("src-pad-added", _src_pad_added, pipeline)
def startme(p):
        p.set_state(gst.STATE_PLAYING)
gobject.idle_add (startme, pipeline)

loop.run()
```

Demo

- 3 way conference ...
- Oops ???

The Future

- Complete RTP implementation
  - DTMF
  - Confort noise
  - sRTP
  - Stabilize
- Port Telepathy to use it
- Use it in all Free clients so they can gain AV capabilities

Thank you

- Farsight is brought to you by Collabora
  - Phillippe Kalaf
  - Youness Alaoui
  - Olivier Crête
- Questions?

http://farsight.freedesktop.org/

http://git.collabora.co.uk/

http://www.collabora.co.uk/

#farsight @ freenode

Collabora